BEHRINGER

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The ProfessionalAudio SourceBook

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KS-T2000

Compact 3-channel Mixer

The KS-T2000 is a compact 3-channel mixer that is small and light enough to go anywhere. This product is made and guaranteed by Kamesan and is ideal for all audio applications in professional broadcast and film work. Camcorders are shrinking but that doesn't mean audio requirements are getting any less demanding. Despite being one-third smaller and about one pound lighter than any comparable mixer, the Kamesan KS-T2000 offers all the flexibility you'll need to engineer precise location sound. True 'Pack and Go' design. At first sight, a 3-in / 2-out configuration does not appear to be a very comprehensive mixer specification. But on location, ease-of-use and adaptability count as the most valued features. You need to interface with varied signal levels, provide multiple feeds, and cope with awkward working conditions. Attention to smallest design details in the KS-T2000 has resulted in a mixer that can be relied on to excel in every situation.

FEATURES

Accommodating Inputs

- 3 switchable XLR-balanced mic/line inputs with 48v and A-B (12V) phantom powering.
- A variable input sensitivity control adjusts the input amplifier gain to optimize its amplification for any level, from a hot condenser to a low output ribbon mic.
- Inputs will match a wide variation in line signal levels. This means that the normal physical position of the main channel faders can be 'trimmed' to be at 2/3 rotation – providing adequate rotation range to 'ride' actual variations in signal levels.
- Switchable 160Hz high pass filters eliminate most low-frequency noise problems.
- Separate left / right output switches route the signal to the main faders and outputs.

Versatile Outputs

- To achieve such a compact size, a single meter is provided though the mixer has full stereo capability.
- The transformer-balanced XLR outputs, can drive at a professional line level (+4dBm) or can be switched down to a low mic signal level. (-60dBm).
- Two RCA connectors for connection to a consumer camcorder's mic input.
- Low channel crosstalk and output isolation allow the mixer to be configured for multiple feeds as the situation demands.
- 1kHz tone oscillator provides a useful calibration signal.

Versatile Monitoring

- Both headphone output and metering can be switched to left or right outputs, or to a mono mix of both. When used for multi-mono applications, one of the main outputs can be designated as a separate live feed and monitored independently.

Exceptional Ease Of Use

- The side panel reveals just how convenient operation is. Only the three main and monitor faders have tall knobs. It is virtually impossible to 'knock' any of the other controls inadvertently during operation.
- Even the main faders have been located on the side of the mixer allowing the operator to set and forget – a useful feature when inexperienced operators use the mixer.
- Wide knob spacing means that the channel fader provides for an easy grip under all conditions.
- An all-weather case allows the mixer to be slung around the neck or shoulder and operated beneath a transparent cover under all conditions.
- The KS-T2000 runs five hours on four AA batteries. Higher voltages can provide higher output levels.
- Alternatively, the KS-T2000 will accept and regulate any external DC supply in the range of 10 to 15-volts.

KS-107 *Lip Checker*

A unique product, the KS-107 helps to track and eliminate broadcast signal time differences between audio and image signals and also multiple audio signals. At the receiving site, audio is easily captured via a microphone or direct line connection. Video bursts are detected directly from a monitor screen using an exclusive image probe, which is manually pointed at the flashing image on the monitor. The calculated delay is displayed on the unit's large LCD to an accuracy of within one millisecond, allowing the operator to make necessary adjustments.

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KS-342

4-Channel Compact Mixer

The KS-342 is a no compromise high-performance mixer that adapts to your needs—since no two location projects are ever the same, and often, they have demands beyond the traditional sound mixing techniques. Ideal for all audio application in professional broadcast and film work.

FEATURES

Input Facilities

- Selectable mic/line inputs providing 12- or 48-volt phantom power as required.
- Level trim at both sensitivities means the main channel faders can be set at optimum rotation for normal working.
- Massive 30dB pre-fader headroom plus continuously variable hi-pass filtering ensure signals never get out of hand.
- Channels can be ganged for true stereo operation and switched to the left or right mix busses.

Outputs

- Thumb-knob master level controls prevent accidental adjustment and the signal passes through switchable compressors before the final output stages.
- Fast attack, compound release circuitry provides a comfortable safety margin for distortion free operation.
- Independent as well as ganged settings ensure maximum flexibility for stereo and dual-mono setups. This design philosophy is carried through to the isolated outputs, which are not a luxury but a necessity in many situations. With a choice of levels (left/right independent), symmetry and connectors, four isolated two-channel feeds are provided.

Comprehensive Monitoring

- Backlit, jam-proof, level meters offer a clear visual indication of the main output signal.
- The headphone circuit can be switched to monitor any combination of the outputs in mono or stereo.
- Additionally, an indispensable PFL function, which appears on the right meter as well as at the headphone output, is included.

Small and Light and Expandable

- Smaller and lighter than any comparably featured mixer, the front and side panels are uncluttered and easy to use.
- The most-important channel faders feature latex knobs for all-temperature comfort and are easily gripped, even in gloves. Kamesan's trademark bar-grip knobs are used for all presets and never obstruct easy operation.
- Runs 8 hours on NP-1 battery in a quick-change compartment, which also accepts a battery pack using 8 AA batteries. Plus up to 16v DC can be applied to the mixer power circuit, which also provides an auxiliary regulated output for powering receivers, etc.
- The nature of location recording is that there will always be situations that require more channels or more processing. The KS-342 features a multi-way link socket that can be used to cascade a stereo input from another mixer without sacrificing channels.
- Secondly, a discrete connector and sturdy locks attach and integrate a range of specialized modules that expand capabilities instantly. Currently two variations are available; the KS-6001, a four channel sub mixer with identical features, and the KS-6002, a block of four parametric equalizers and compressors.

	KS-342	KS-T2000
INPUT		
Mic Input Level (3kΩ)	-70 ~-30dBm (Balanced)	-70 ~-30dBm (Balanced)
Line Input Level (600Ω)	-20 ~ +4dBm (Balanced)	-20 ~ +4dBm (Balanced)
Aux Input Level (10kΩ)	-20 ~ +4dBs (Balanced)	
Ext Mon Input Level		0dBs/600Ω x 2
Low Cut Filter	20 to 200Hz (Cont. Var/12dB/octave)	160Hz (12dB/octave)
Headroom	> 34dB (Pre-fader input)	> 30dB (Pre-fader input)
OUTPUT: Output level		
Main L/R (600Ω, Trans. Output)	Selectable +4/0/-20/-60dBm	Selectable +4/-60dBm
Sub L/R (600Ω Load, Elec. Bal.)	Selectable -20/-60dBm	–
Unbalanced Output	-60dBs (>10kΩ at full load)	-60dBs (600Ω ~10kΩ)
Monitor(Max Load, 8Ω, Stereo)	0dBs/50Ω	-10dBs/50Ω
OUTPUT: Maximum Output		
Main (600Ω Load, +4)	+24dBm (at 12V), +20dBm (at 8V)	+23dBm (at 6V)
Sub L/R (600Ω Load, -20)	+8dBm (at 12V)	
Monitor(50Ω Load)	+6dBs (at 12V)	+5dBs (at 6V)
AES/EBU Digital Output	Selectable 44.1/48/96 kHz (20 bit A-to-D)	
Compressor	No	Yes
AUDIO Frequency Response	30Hz~40kHz ±1.0dB	50Hz~15kHz ±0.5dB
Distortion (T.H.D)	<0.1% (+4dBm output 50Hz~15kHz)	<0.1% (+4dBm output 50Hz~15kHz)
POWER: Internal Battery	'AA' x 8 (BP-3/8 Case Included) NP-1 NiCd Battery (opt. Sigma case)	'AA' x 4 (Holder Included)
Operating Time (Continuous)	>8 hrs (Alkaline Batteries)	>8 hrs (Alkaline Batteries at 77°F)
Dimensions (WDH)	8.46 x 6.99 x 2.64″	6.26 x 5.67 x 1.85″
Weight (no case or batteries)	Approx 4.4 lbs.	Approx 2.2 lbs.

ALPHAMIX

Portable 4-Channel Audio Mixer

The AlphaMix contains all of the features required by today's most demanding ENG recordist. Do you regularly do two camera shoots, work with demanding producers or have complicated IFB feeds? No problem. The AlphaMix features four switchable mic/line inputs, 12T or 48v phantom power, low noise pre-amplifiers with continuously variable gain trims, active low cut filters, balanced line level outputs per input channel, pre-fader-listens and simple three LED meters per input channel.

The AlphaMix uses NP-1 style batteries and is equipped with selectable low battery monitoring. In addition, it is equipped with four Hirose 4-pin power output connectors that allow it to power up to four wireless receivers. And unlike other external battery distribution boxes, each of the power outputs is individually filtered for quiet wireless operation. The AlphaMix was also designed from day one to feed and monitor audio returns from two cameras. Outputs include special, custom made, dual isolated output transformers that are the largest in the industry. Finally, the AlphaMix is the only mixer to offer two-camera support, and it utilizes a custom-made peak reading LCD meter that emulates most of today's cameras. For further convenience, it is equipped with a boom pole mounted remote control that can be used to control the gain of the first input channel, thus allowing you to boom with both hands.

- Four line or mic level switchable inputs
- 12T and 48v phantom powering
- Active 12dB/octave low cut filters.
- Retractable gain trims on each pre-amp
- Retractable pan pots
- Individual balanced outputs per input channel
- Individual 3 LED meters per input channel
- Pre-fade listens per input channel
- Boom pole mounted remote control slide fader allows booming with two hands
- Custom made (properly sized) dual isolated output transformers
- Two Hirose 10 pin connectors for dual camera support, line/mic level switchable
- Two 3.5mm stereo auxiliary outputs, line or microphone level switchable
- Two XLR balanced outputs
- Output limiters, switchable separate or ganged
- Additional output that provides balanced mono feed
- Built in 4-way power distribution for wireless receivers
- Built in slate microphone
- Built in reference tone oscillator
- Built in battery monitoring, switchable calibration for all NP-1 battery types
- Switchable headphone monitoring left, stereo, right, mono, MS as well as tape returns from 2 cameras.
- Aircraft aluminum housing with epoxy powder coat finish.
- Lexan overlays with wear-proof subsurface silk-screening.
- Includes Porta Brace case and strap

MJR *2-Channel Mixer*

Perfect for use with today's small DV camcorders, DAT and Minidisc recorders, the PSC MJR Mixer offers the features of full size mixer in a small, portable format. The MJR Mixer contains two high quality mic preamps, each with switchable line or mic inputs, high pass filters, channel assignment pan switches, and 48v phantom power. It also offers a slate microphone, reference tone oscillator, headphone amplifier with tape/direct switch, sunlight readable LED meters and a multi-pin connector for use with PSC's BetaSnake cables. All of this housed in a rugged aircraft aluminum case barely 4.75 x 6˝ and weighing less than 1.5 lbs. In addition, the entire mixer can be mounted to the bottom of a small video camera via the supplied 1/4-20 thumb screw. And the whole mixer/camera combination can be mounted to any standard tripod as well via 1/4-20 threaded attachment point. Runs on a single 9-volt battery for more than five hours.

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M3

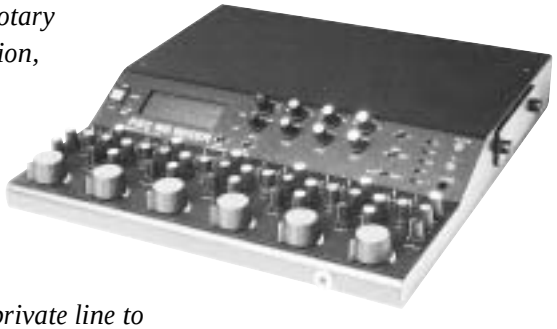
3-Channel ENG Audio Mixer

A cost effective, flexible portable mixer that combines rugged construction, high quality componentry, and ease of use ergonomics in a compact package, the M3 is perfect for ENG and EFP applications, digital and desktop recording and editing. The M3 features 3 high-quality, transformer-balanced XLR inputs and 2 (L/R) stereo paired XLR outputs capable of both mic and line level operation. Output limiters eliminate transient peaking caused by signal overload. The dual limiters can either be used independently or "gang" them together for applying the limiter to stereo signals. The M3 is powered via AA batteries offers 48-volt phantom power, and has a dual multi-pin connector for a single interface between the M3 and Betacam cameras. A convenient on board slate microphone is standard and a supplied boom pole remote control allows engineers the freedom of adjusting record levels while handling the boom pole.

- Three high quality transformer-balanced XLR inputs (with switchable "T", Dynamic or 48v phantom powering) and outputs for pristine audio with low noise. All inputs and outputs are mic/line switchable and have a 3-way input pad: Dynamic 0dB Pad, Condenser -15dB Pad, Line Level -50dB Pad.
- Additional outputs include two 10-pin Hirose connectors for use with Beta snake cables (mic or line switchable) for conveniently connecting to two cameras, and two Auxiliary outputs for transcription or RF feeds (mic or line switchable). The 10-pin connector provides audio signal flow as well as tape return signal flow for monitoring.
- Ganging input and output connectors allow you to use two M3 mixers tied together.
- Channels 3 and 4 can be linked for stereo recording using 2 mics in either M-S or X-Y configurations. The "Gang" switch links inputs 3 and 4 and both input levels are then controlled by Pot 4.
- "Separate" and "Ganged" switches allow independent or linked operation of both limiters.
- The LCD Peak meters emulate the peak reading attributes found in many Betacams. This ensures the meters accuracy no matter how the mixer and camera are connected.
- Low cut filters eliminate low frequency noise and rumble.
- The momentary "Slate" button activates a built in slate microphone for recording notes and information related to editing.
- Headphone monitoring selections include Left to both ears, Stereo, Right to both ears, Mono, Pre Fade Listen of channels 1,2 and 3 and MS Stereo as well as Tape returns from two Betacams.
- A handy boom-pole remote control attaches to any boom pole. This allows sound engineers to adjust recording level without reaching for the mixer.

M6

6-Channel Portable Mixer for Location Sound Recording



The M6 Mixer was designed for extreme field operating conditions (utilizes rotary pot faders because of their immunity to dirty and dusty conditions). In addition, it offers provisions for direct mounting of a DAT or DVD recorder. This allows for a simple and convenient means of recording in the field whether you are working on a sound stage, in an insert car, or at some other remote location.

The M6 Mixer offers six inputs each with switchable mic powering, line or mic inputs, phase reversal, 2-way EQ, pan pot and two aux sends. Its output section contains two large, easy to read LCD peak meters, slate microphone, private line to boom operators, full duplex boom communication, recorder remote rolls, reference oscillator, four selectable headphone feeds, two main outputs, two aux outputs, and a selectable video assist output.

DMX-P01

Digital Portable Field Mixer

Effective audio quality is now more important than ever for ENG and EFP applications, because even with high-quality video, lackluster audio performance will result in a mediocre production. The DMX-P01 is Sony's answer to these high-quality audio requirements. Not only does it offer outstanding audio quality, it is also one of the most user-friendly mixers in the ENG and EFP markets. With full 24-bit processing and 48kHz or 96kHz sampling rate, the DMX-P01 provides sound quality comparable to high-end production mixers. Its front panel is engineered to allow fast, easy, and accurate setting adjustments - essential when working in the field. It also includes other useful features, such as panel-lock and parameter-lock functions, selectable meter scales, camera-audio return-level check, memory function, digital cascade capability, and digital outputs. All of this functionality and versatility is packaged into one sleek, compact, and lightweight body that can be used effortlessly in field productions.

Digital Benefits of the DMX-P01

Excellent Sound Quality

- In order to provide outstanding sound quality, the DMX-P01 offers full digital-audio processing. Its 24-bit A/D and D/A converters provide a high level of linearity for analog inputs and outputs. Internally, the unit utilizes 32-bit digital processing for maximum throughput. In addition, the DMX-P01 has a sampling rate that is selectable from either 48kHz or 96kHz.

Digital Limiters/Compressors

- By using digital limiters and compressors, the DMX-P01 can provide extremely high-quality sound in one small package. Limiters are available at the input, and both limiters and compressors are available at the output for maximum flexibility.

Full Parameter Controls

- The DMX-P01's front panel puts all of its controls at your fingertips with an organized and logical layout. The DMX-P01 processes audio digitally, so parameters that are used less frequently are stored internally and accessed only when needed. Using the front-panel controls and easy-to-read backlit LCD allows full control of every parameter without the need to remove the unit from its audio-organizer case.

Panel-Lock

- One major concern for engineers in the field is the accidental bumping of controls, which can change the sound settings. The DMX-P01 safeguards against this with its panel-lock feature, which can be set to secure all of the control settings, or selected individual control settings.
- In addition, a Parameter-Lock feature also avoids inadvertent parameter changes.

Flexible Meter Scales

- Being digital, meter calibrations can be easily changed from one type to another without the need to replace the entire meter. Six easy-to-change meter scale sheets are supplied:
  VU, PPM1 (BBC-type), PPM2 (DIN-type), PPM3, PPM4 (IEC-type1) and dBFS.
  Simply insert the desired scale sheet and select the right meter type. The DMX-P01 will display the audio level according to the scale selected.

Camera-Audio Return Check

- The DMX-P01 enables users to visually verify that the mixer's audio level matches the level recorded to the camcorder tape. This is done using the camera return-level mode, available in the setup menu.

Digital Output

- The DMX-P01 is equipped with a digital output, which can be used to send audio to digital equipment such as DAT recorders. AES/EBU and S/PDIF coaxial interfaces are available.

Memory Function

- Easily store and recall parameters from the setup menu. There are two memory functions:
  – Power-On Memory Recall - When the DMX-P01 is powered on, the system can recall parameters in three ways: default factory settings, the last settings used, or with the parameters of one specific scene memory.
  – Scene Memory Recall - This allows users to recall up to ten different user-defined parameter settings or the factory default settings. Invaluable in situations where a single unit is required to serve multiple users or multiple shooting scenarios

Digital Cascade

- When additional inputs are required, DMX-P01 mixers can be cascaded using a digital connection between mixers. Best of all. sound quality isn't degraded, as it would be when cascading analog mixers.

DMX-P01

Side views of the DMX-P01

Inputs

- Four XLR-balanced mic/line inputs
- +48 V power for each mic input
- Digital cascade input with phono connector
- Microphone/line gain-level control
- Level control knobs with stereo-link facility
- Selectable sampling rate (48 kHz or 96 kHz) for A/D converters

Outputs

- 2 balanced outputs on XLR-type balanced connectors
- Digital AES/EBU output (stereo) on XLR-type balanced connector
- Coaxial output connector for mix-bus output (for cascade) or S/PDIF digital output (selectable)
- Stereo tape output on unbalanced 3.5 mm TRS jack
- Switchable output mode: stereo or monaural
- Selectable output-level control for L/R master outputs and camera send
- Selectable sampling rate (48 kHz or 96 kHz) for D/A converters

Panning

- Variable pan controls

Low Cut Filters

- Adjustable (50 to 200 Hz) cut-off frequencies for 2 user settings (A/B)
- Quick parameter-recall switch with OFF/A/B positions

Limiters/Compressors

- Digital limiters on both inputs and outputs
- Digital compressors on outputs
- Precise parameter control on threshold and ratio value, attack and release time
- Link function (ON/OFF switchable)
- LED indicators for output limiter/compressor operation

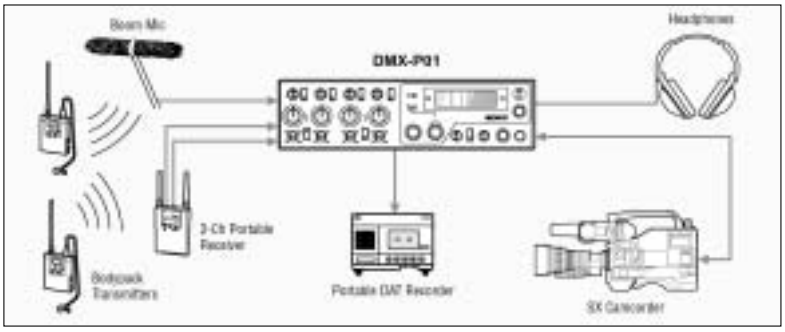
Link/M-S Operation

- Links input levels, LCFs, and PAN controls for channels 1/2 and 3/4
- Links output levels and limiter/compressor settings for master left/right outputs
- Decodes M-S microphone inputs, and links the input levels of channels 1/2 and 3/4
- Phase reverse on channels 2 and 4 (M-S decode)

LCD Panel

- Various level-meter displays: VU, PPM1 (BBC-type), PPM2 (DIN-type), PPM3 (NORDIC-type), PPM4 (IEC-type1), dBFS
- Displays setup menus and allows various parameter settings
- Three quick-recall memory settings for immediate mixer setup
- Ten user-scene memory settings (each including level meter, LCF, limiter/compressor, and link status)
- Six scale sheets supplied for different level-meter calibrations
- Back light
- Heated LCD for low-temperature conditions

SYSTEM EXAMPLE



Monitoring

- 2 outputs: 1/4″ phone jack and 3.5-mm mini jack
- Six monitoring modes: left output, right output, stereo output, left/right-mixed monaural, M/S decode and camera return
- Level-control knob

Camera-Audio Send/Return-Level Control

- Stereo return from a camcorder via 12-pin Tajimi balanced connector
- Precise level control on LED with auto-evaluation function for return level
- Monitoring capability with headphones

Oscillator/Talkback (selectable)

- Oscillator: 1-kHz pilot-tone signal into all outputs
- Talkback: slate into all outputs
- Momentary and alternative modes for both oscillator and talkback

Power

- External DC 10 to 15v input with 4-pin XLR connector
- External DC 12 V input with jack connector
- DC 12 V output on 4-pin Hirose connector
- 8 internal AA-size (LR6) alkaline batteries for 5 hours of continuous operation

SHURE

FP-33

Portable 3-Channel Stereo Mixer

The standard by which portable mixers are measured, the FP-33 is used all over the world for remote audio recording, ENG/EFP applications and location film production. Built upon the benchmark FP32 and FP32A field mixers, the FP33 is light enough, small enough and rugged enough to take anywhere. It features three XLR-balanced mic/line inputs and two outputs, center detented pan pot for each input channel, oscillator, two headphone jacks, 48v/12v phantom and 12v T (A-B) power, tape out jack, and a monitor input for the headphone circuit. Exceptional low noise design and wide dynamic range make the FP-33 ideal for use with digital transmission links or digital video/audio media, including DAT and recordable CD.

FEATURES

Inputs

- Three XLR-balanced inputs; switchable to low-impedance mic or line level.
- Phantom or A-B (T) power for condenser mics is available at each mic input.
- Built~in tone oscillator for level checks or line tests.
- Slate microphone with automatic gain control (AGC) for take identification or for emergency use. Slate tone for identifying take locations during editing.
- Stereo monitor input allows headphone monitoring of external sources without interruption of mixer functions.
- Link switch couples mixer inputs 2 and 3 into stereo pair

Outputs

- Left and right XLR-balanced outputs are mic/line switchable.
- Left and right tape outputs are available to feed tape recorder inputs or other unbalanced aux-level inputs.
- 1/4″ and 3.5 mm headphone jacks are driven by a stereo headphone power amp with separate level control.

Controls and Indicators

- Center detented pop-up pan pots on each input. Color-coded, soft touch rubberized knobs with tactile position indicators
- Active, feedback~type input gain controls permit direct input of high-level sources without input attenuators.
- Dual clutched Master gain control for individually adjusting left and right levels at line/mic and tape outputs, as well as tone oscillator and slate mic levels.
- 150Hz (6dB/octave) Lo-cut filters at each input reduces extraneous low~frequency interference.
- Built-in limiter with adjustable threshold prevents output clipping of mixer or input overload of amp or tape deck.
- LED indication of input level, peak output level, limiter action, and battery.
- Illuminated VU meters for left/right channels. Preset for 0 VU = +4 dB, adjustable for other levels.
- Headphone monitoring mode switch and headphone MS matrix.

Performance

- Wide, flat response with extremely low distortion and up to +1dBm output level for studio-quality performance.
- Small and lightweight, the FP-33 offers sealed input potentiometers and a steel chassis, making it extremely rugged and durable as well.
- Mix bus jack for connecting additional FP33 or FP32A mixers
- Internal DIP switches provide over 4,000 different set-ups

Power

- Powered by two standard 9v batteries (up to 8-hours under normal conditions) that can also supply 48v or 12v phantom power to condenser microphones.
- Can also be externally powered from any 11 to 30v DC source such as battery belt pack or car battery.
- A third 9v battery can be used for condenser mics that require A-B power.

FP-33 with carrying case, shoulder strap and mix bus cable.....**1199.95**

Applications Include:

◆ Remote audio recording ◆ Electronic field production (EFP) ◆ Electronic news gathering (ENG) ◆ Film production

FP-410

Portable Automatic 4-Channel Mic Mixer with IntelliMix

A portable automatic mixer, the FP-410 is equipped with Shure's patented "IntelliMix" to deliver flawless automatic microphone mixing. The remarkable operating concept behind the FP-410, "Intellimix" combines three unique functions: Noise-Adaptive Threshold, MaxBus and Last Mic Lock-On, to provide greater gain before feedback, reduce pickup of ambient noise and eliminate comb filtering effects. Ideal for video production, broadcast, conference recording and field production, the FP-410 is also useful for convention facilities, hotels and sound installations.

Noise-Adaptive Threshold:

Distinguishes between constant background noise (such as air-conditioning) and rapidly changing sound (such as speech). This function continuously adjusts the activation threshold so only speech levels that are louder than background noise will activate an FP-410 channel.

MaxBus:

Eliminates the poor audio quality that results when a speaker is picked up by more than one microphone. It does this by controlling the number of microphones that may be activated for a single sound source. With MaxBus, one talker will activate only one FP-410 channel, even if multiple microphones are "hearing" that talker.

Last Mic Lock-On:

Maintains a seamless audio mix by keeping the most recently activated microphone open until a newly activated mic takes its place. Without this function, a long pause in conversation might cause all mics to turn off and sound as if the audio signal has been lost. With Last Mic Lock-On, background ambience is always present.

"IntelliMix"—How it Works

Multiple miking situations—with a number of talkers participating—have always presented problems for the audio technician. If too few mics are used, the coverage of each talker may vary, with one talker (nearest the mic) being louder and clearer than the next. Talkers farthest from the mics will sound "echoey" and reverberant, as very little of the direct sound from their mouths reaches the microphones. If too many mics are used, there's more background noise and reverberation pickup, as well as less gain before feedback if a sound reinforcement (PA) system is used.

It's somewhat like having multiple video cameras all focused on the same subject. If the camera signals are combined, the result is a blurred image. When multiple microphones are open for a single talker, the result is a blurred audio signal. But it's often not practical for someone to turn mics on when they are needed and off when they are not. The answer is the FP-410.

The FP-410 automatically attenuates any microphone not being used, greatly reducing excess reverberation and feedback problems. When a new talker starts, the FP-410 immediately selects and silently activates the most appropriate microphone. "IntelliMix" electronic processing enables the FP-410 to provide clear, natural voice pickup. The FP-410 significantly reduces the problems of "boomy" or "muddy" sound, insufficient sound level (because of feedback or "howling"), and operator errors. In fact, errors are virtually eliminated because the FP-410 doesn't need an operator or technician for continual adjustment—once set up, it is completely self-sufficient.

The FP-410 has numerous applications in video production and audio recording, broadcasting, and sound reinforcement. In any speech pickup application with multiple microphones, the FP-410 dramatically improves audio quality. Switching from manual to automatic operation allows an individual's voice to rise above background noise and reverberation to become clearer and more intelligible.

- Four XLR-balanced inputs and two XLR-balanced outputs. Each can be individually set for microphone or line-level signals.
- Any high quality, low-impedance, balanced mic (dynamic or condenser) can be used, including wireless and shotguns
- Additional FP-410 mixers can be interconnected. Linked systems can contain over 25 mixers and 100 microphones.
- Front-panel channel gain and master controls operate as in conventional mixers.
- Selectable hold time keeps microphones on during short pauses in speech.
- Automatic gain adjusts as additional microphones are activated.
- Wide, flat frequency response and low distortion up to +18 dB output.
- LED indication of mic channel mix levels, output level and limiter action.
- Automatic muting prevents annoying thumps and loudspeaker damage when the FP-410 is turned on and off.
- Separate monitor input and tape output (aux-level) jacks.
- Front panel headphone monitor jacks with level control.
- 48-volt phantom powering for condenser microphones.
- Operates on AC or two 9-volt batteries.
- Includes optional bumpers (feet) for use on horizontal surfaces; a short cable for linking two FP-410's; and a rack mount kit for installation in a 19˝ rack.

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SHURE

FP-24

Portable 2-Channel Stereo Preamp and Mixer

The FP-24 is a studio-quality, two-channel, portable stereo microphone mixer, designed for active use in demanding broadcast environments. Features including assignable L-C-R inputs, built-in slate microphone, 1kHz tone oscillator, and headphone monitoring render this mixer extremely adaptable in application. The remarkable audio performance and comprehensive attributes of the FP-24 make it an appropriate choice for studio or field production engineers. Due to its compact and rugged mechanical and electrical construction, the FP-24 excels in no-compromise settings such as radio, television and film production.

- Two transformer-balanced microphone inputs with left, center, and right position mixing capabilities
- Unique 7-segment peak output meters, with three selectable levels of LED brightness—readable even in direct sunlight
- High current balanced output drivers provide signal integrity over long cable runs
- Headphone preamplifier enables monitoring of program audio or external tape return
- "Unclippable" input peak limiters with adjustable threshold (each input)
- Selectable 15v and 48v phantom power
- Switchable low-cut filters with 80 and 160 Hz corner frequencies, 6dB per octave
- Powered with two AA alkaline batteries for 11-12 hours) or 4-14v DC power (PS20)
- The FP-24 has the ability to link with the Shure FP33 mixer using the optional A33LK cable kit to create a highly portable, flexible, and cost-effective 5 x 2 audio production system. (The A33LK kit can also be used with the Shure FP23 to create a 4 X 2 setup.)

M367

Portable 6-Channel Mic/Line Mixer

An industry standard, the M367 is a six-input mono mic/line mixer/remote preamplifier specifically designed for professional applications. A complete and compact console, the M367's excellent performance, versatility and features make it ideal for studio, remote, video deposition and sound reinforcement applications as well as an add-on mixer for expanding existing facilities. Built to meet the requirements of the most demanding field production applications.

- Six switchable XLR-balanced mic or line level inputs with individual gain controls and low-frequency roll-off switches
- Two XLR-balanced outputs; one selectable mic/line output and one dedicated line output
- Metal XLR connectors on both inputs and outputs, detachable AC cable
- Feedback-type input gain controls for maximum clipping levels and dynamic range
- Built-in switchable peak limiter cuts output overload distortion and adapts to power supply voltage
- LED indicator shows limiter operation or overload with limiter defeated
- Externally adjustable limiter threshold (-4 to +18dB)
- Wide, flat frequency response (20Hz to 20 kHz) and extremely low distortion up to +16dBm line level output
- VU meter is calibrated for +4 and +8dB with range switch. Meter is also illuminated during AC operation
- Phantom power for condenser mic operation
- Front-panel headphone level control and monitor jack; can drive almost any stereo or mono headphones
- Headphone output level is high enough to be used as an auxiliary unbalanced line feed to drive a tape recorder or power amplifier
- Automatic muting prevents speaker damage during power on/off
- Highly stable, low-distortion tone oscillator provides for line test and level checks
- Rear panel Mix Bus jack facilitates stacking multiple M367's for additional input capability without losing any inputs. Two M367's connected, provides two independent master gain controls and two isolated line amplifiers with eight individually controlled inputs
- Selectable 120 or 240 v AC operation as well as portable DC capability (three 9v batteries required)

M367 .....**524.50**

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EQUIPMENT LEASING AVAILABLE

FP-42

Portable 4-Channel Stereo Mic Mixer

The FP-42 is a four-input, two-output, compact self-contained stereo mixer with all the features and ruggedness that have made Shure mixers the industry standard. Perfect for mixdown in video editing suites, it integrates all the operating features and reliability of a professional studio mixer in a single unit that is small and light enough for location use.

- Four XLR-balanced inputs, each mic/line switchable with low-cut filters and cueing options
- Left and right XLR-balanced outputs with mic/line and mono/stereo switches
- Active feedback-type input gain controls for high level signals without input attenuators
- Pull-pot cueing on all inputs provides channel previewing
- Built-in tone oscillator permits level checking and line testing
- Parallel stereo headphone jacks (1/4″ and mini) with level control
- Adjustable-threshold limiter with left and right channel peak indicators
- VU meters with range switch and battery check function
- Left and right channel master level controls and ganged headphone level control
- Powered by three 9v batteries or via internally selectable) 120 or 240v AC power

	M367	FP-33	FP-42	FP-410	FP-24
Transformer-Balanced Inputs and Outputs	Yes	Yes	Yes	Yes	Yes
Active-Balanced Outputs	—	—	—	—	Yes
Low-Z Mic-Level Input	Yes	Yes	Yes	Yes	Yes
Line Level Input	Yes	Yes	Yes	Yes	—
Aux Level Input	—	—	—	—	Yes
Mic Level Output	Yes	Yes	Yes	Yes	—
Line Level Output	Yes	Yes	Yes	Yes	Yes
Phono Jack Aux Level Output	—	—	—	—	Yes
Headphone Output	Yes	Yes	Yes	Yes	Yes
Phantom Power	48v	48v	Yes	48v	Yes
VU Meter	Yes	Yes	Yes	Yes	—
Peak Meter	—	—	—	Yes	Yes
EQ	Yes	Yes	Yes	Yes	Yes
Tone Oscillator	Yes	Yes	Yes	Yes	Yes
Linkable	Yes	Yes	Yes	Yes	To FP33
Slate Mic & Tone	—	Yes	—	—	Yes
Limiter	Yes	Yes	Yes	Yes	Yes
Stereo Operation	—	Yes	Yes	—	Yes
Power Options	AC/DC	DC Only	AC/DC	AC/DC	DC Only
TYPICAL APPLICATIONS					
Automatic Microphone Activation	—	—	—	Yes	—
Church Sound	Yes	—	—	—	—
Broadcast	Yes	Yes	Yes	Yes	Yes
Field ENG/EFP	Yes	Yes	Yes	Yes	Yes
Basic Microphone Mixing, Basic PA	Yes	—	—	—	—
Distance Learning	Yes	—	—	Yes	—
Video Editing Suite	Yes	—	Yes	—	Yes
Conference Rooms, Round Table Discussion	Yes	—	—	Yes	—

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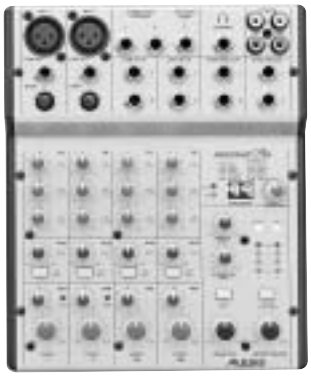
MULTIMIX SERIES

Alesis' MultiMix series is a line of powerful, portable, affordable mixers designed for a wide range of operating environments and applications. Available in 4 different tabletop mixers, plus a powerful rack mixer, all feature the high quality and high performance that Alesis is known for. Each mixer features 99 on-board 24-bit Alesis digital effects, for incredible control and creativity, as well as an intuitive interface design with superior feel. And each mixer has been designed from the inside out to provide the level of performance users expect from Alesis, at incredible prices.



MultiMix 6FX

A super compact analog mixer for performing or recording applications, with digital effects. An incredible value (under $100), the MultiMix 6FX includes 2 mic/line inputs, 2 stereo line inputs, a 3-band EQ on each channel, plus 99 high quality 24-bit Alesis digital effects.

MultiMix 8FX

A highly compact analog mixer, designed for live applications such as piano bar, single musician, duets, etc., or for simple recording applications such as live to 2-track, etc. Delivering powerful mixing capabilities at a very low price, the MultiMix 8 FX includes 99 high quality 24-bit Alesis digital effects, and is small enough to fit in a "gig bag." Featuring 8-input, 2-buss analog mixing with 4 mic preamps with switchable 75Hz high-pass filters, 2 stereo line channels, a 3 band EQ on each channel, and more.

MultiMix 12FX

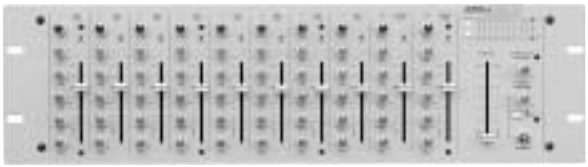
A portable analog mixer that can handle a wide range of PA and recording applications, such as live performances for small bands, ensembles, and general PA, or recording applications such as live to 2-track, simple multi-track, etc. The low cost, yet powerful MultiMix 12FX includes a 12-input, 4-buss analog mixer featuring 4 mic preamps with switchable 75Hz high-pass filters, 60mm linear faders on each channel, 4 stereo line channels, a 3-band EQ on each channel, 99 high quality 24-bit Alesis digital effects, and much more.

MultiMix 12FX D

The MultiMix 12FXD is the same as the MultiMix 12FX plus it adds a 44.1kHz digital S/PDIF digital output—making it ideal for those seeking a low-cost, yet powerful mixing solution with digital output.

MultiMix 12R

The world's most affordable 12-input, 2-buss analog rack mixer featuring 8 mic preamps with 2 stereo line channels, all in a compact, 3-U rack mount configuration. The MultiMix 12R has 2-band, fixed frequency EQ on each channel, 1 pre-fader aux send and 1 post-fader aux send per channel, and 60mm linear potentiometers (faders) for master level of each channel. The master section features an external stereo aux return level, stereo LED bar graph meters, stereo master L/R buss 60mm fader, and a separate phones/monitor level control.



- 12-channel analog mixer in compact 3U rackmount design
- 8 XLR mic inputs with phantom power
- 2 aux sends (pre-and post-fader) for monitor returns and effects
- Insert points on 8 channels, plus 2 channels of stereo line inputs
- 2-band shelving EQ at 80Hz and 12 kHz
- 60mm linear faders on each channel

M-3 TUBETRACKER

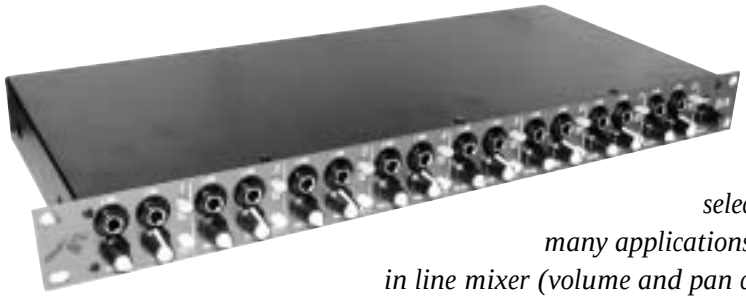
8/2 Tube Mixer

The M-3 is an 8-channel compact tube mixer, ideal for all tracking and mixing applications. Each channel features discrete tube mic preamps, a 4-band EQ with swept mids, 2 aux sends and a post-fader direct output for multitrack recording. The master section has tube stages in the main mix bus, together with 2 aux returns, monitoring, metering and an optional 24-bit/96kHz stereo digital output. When linking multiple M-3 mixers, the stereo, PFL and aux busses are also linked, providing an unlimited number of channels.

- Balanced XLR mic and 1/4" jack line inputs
- Balanced XLR L/R mix outputs
- Optional AES, S/PDIF and TOSLINK optical digital outputs
- Sovtek ECC83/12AX7A tubes
- Phantom power, 90Hz filter, phase reverse and mic/line switch
- 100mm faders with pan, mute and solo
- Drive and Peak LEDs
- Balanced internal signal busses
- Stereo main fader with VU metering
- Unbalanced 1/4˝ jack inserts
- Word Clock input
- Switchable level direct outputs
- Link facility via 15-way D-type connectors
- Headphone output
- External linear PSU
- 200v DC stabilized supply
- 19˝rack-mountable or table-top use

SM PRO AUDIO DI4v/DI8

4- and 8-Channel DI/Line Mixers



Multi-function direct injection boxes and stereo line mixers in one single unit rackmount chassis, the half-rack SM DI4v and full-rack SM DI8 feature 4 and 8 channels of unbalanced to balanced converters with 10dB pad and earth lift selector provided per channel. They are suitable for many applications right out of the box but it is the addition of a built in line mixer (volume and pan control for each channel and master volume), that makes them unique problem solvers. If you are a keyboard player, electronic drummer, PA/Studio owner, pro musician or a hobby player, the SM DI4v or SM DI8 will find a place in your rack. Imagine running your 4 keyboard or modules on stage directly onto them and being able to send off balanced signal to the front of house rig while providing your own dedicated monitoring mix...all on a single rackmount direct injection box/line mixer! SM DI4v includes a headphone amplifier so you can monitor your own signal, the SM DI8 includes a LINK function so that multiple units can be linked to a single stereo output. So if you have more than 8-16-24.... inputs of unbalanced signal they can all be routed and monitored through the one output pair!

- 4- and 8-channel DI Box
- 4- and 8-channel Mixer
- Each input has a balanced output
- Earth lift per channel
- 10dB pad per channel
- Volume control per channel
- Pan control per channel
- Master Volume control
- Stereo Line Out
- Headphone Amp (SM DI4v)
- Stereo Link Input (SM DI8)
- 110v-240v external power supply

UB SERIES

Ultra Low-Noise Eurorack Mixers

By continuously responding to the suggestions of their customers, Behringer has developed a range of mixers that offer more bang for the buck without compromising on audio quality. Built to exacting standards, they are rugged and reliable, and offer the flexibility and sonic integrity demanded by the most discerning end users. Ultra-low noise circuitry and transparent audio, balanced mic/line inputs, invisible mic preamps, solid steel chassis, sealed potentiometers make the UB Series Euroracks ready for broadcast production as well as MIDI rigs, home/project studios, video editing suites, and live sound reinforcement. And with ten mixers in the line, there is sure one to meet your requirements.

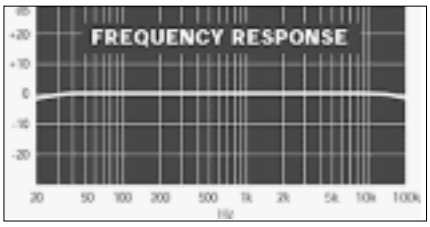
With the UB Series, Behringer takes mixing to the next level. From the miniscule UB502 to the largest UB2442FX-PRO, all mixers combine the same high-quality components and innovative features with affordable pricing. They all feature premium IMPs ("Invisible" Mic Preamps) and ULN (Ultra Low-Noise) circuitry design, and with a feature set of impact-resistant pots and first-rate faders from ALPS, and low-tolerance components, the UB Series provide uncompromising quality under their sleek and sturdy roadproof skin. In addition, the PRO models feature an internal autorange (80-240v) switch-mode power supply and all FX PRO models feature award-winning 24-bit digital and 24-bit/46 kHz internal processing Virtualizer technology with 99 breathtaking stereo effects.

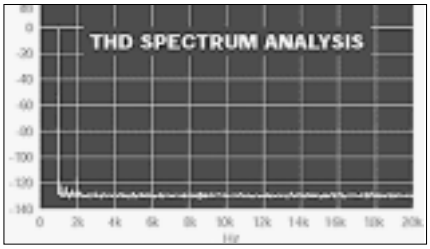
The state-of-the-art, studio-grade IMPs provide 130 dB dynamic range for 24-bit, 192 kHz sampling rate inputs, ultra-wide 60dB gain range and +30dBu line input capacity for crystal-clear audio and ample headroom. With their extreme bandwidth and amazingly neutral, noise- and distortion-free circuitry, the mic preamps deliver crystal-clear audio for awesome sound without any coloration. ULN low impedance circuitry design coupled with high-quality 4580 op amps provides maximum headroom, minimal noise and stunningly transparent audio.

THEY ALL FEATURE:

Invisible Mic Preamp (IMP)

- Five years in the making, the Invisible Mic Preamp (IMP) is a superior mic preamp that delivers crystal-clear audio with incredible reproduction of the slightest nuances. It matches or exceeds the performance of outboard mic preamps. You get extreme bandwidth of 5Hz to 100kHz coupled to unusually neutral, noise- and distortion-free circuitry.





Ultra Low-Noise (ULN)

- ULN stands for the Ultra low-noise design of their circuitry. It is based on extremely low-impedance components that keep thermal inherent noise and crosstalk at an absolute minimum. The result: your console is considerably less noisy and crosstalk between neighboring channels is virtually non-existent.

Switch-Mode Power Supply (SMPS)

- All PRO series mixers (UB1204-PRO and up) use a state-of-the-art, over-sized, switch-mode power supply with lots of headroom. The advantage, compared to conventional circuits, is that a switch-mode power supply automatically adapts to AC voltages between 100 and 240 volts. Furthermore, due to its much greater efficiency, it consumes much less energy than a conventional supply unit.
- Even more important is the fact that conventional transformers always induce 50 Hz hum (check it out on other mixers when you turn up all volume controls). Behringer's switch-mode supply operates at very high frequency and therefore keeps the mixer "dead" quiet.

High-Quality Construction

- High quality ALPS sealed rotary controls and faders keep out dust and moisture to ensure long lasting, reliable performance. Steel chassis and horizontally mounted fiberglass circuit boards are used for their ruggedness, whether on the road or bouncing from studio to studio.

24-bit / 46kHz

- Digital recordings done at 16-bit depths break up the dynamic range between the loudest and quietest portions of a signal into 65,536 steps. Behringer uses 24-bit resolution (dividing the signals' dynamic range into 16,777,216 steps) allowing for far more accurate rendering of the quieter parts of the audio.
- Audio signals change constantly, so a digital device must capture these changes as faithfully as possible. The higher the sampling rate, the better. UB Series mixers use a sampling rate of 46kHz (46,000 times per second). Since the sampling rate should be twice that frequency, 46 kHz is more than adequate for capturing all signals in the audible range—the human ear can not hear sounds whose frequency is higher than 20 kHz.

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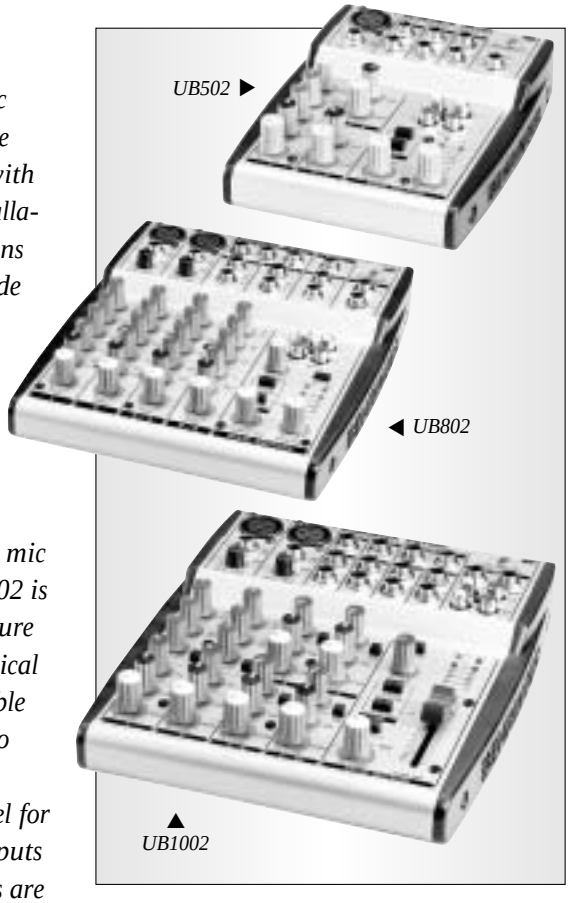
UB502 • UB802 • UB1002 • UB1202

5-, 8, 10- and 12-Input 2-Bus Mixers



It may be small, but the UB502 features the same state-of-the-art Invisible Mic Premap (IMP) and the same Ultra low-noise (ULN) design for highest possible headroom as its larger brethren. Its one mono channel plus 2 stereo channels with a 2-band EQ on the mono channel make it ideal for use with fixed audio installations as well as keyboards, samplers and computer application. Hobby musicians and video makers will also find lots of use for this ultra-compact mixer. A shade larger than the UB502, the UB802 features 2 mic preamps, 6 balanced high-headroom line inputs, 1 post fader FX send per channel for external FX devices and 1 stereo aux return for FX applications or as separate stereo input. There is also an extremely musical 3-band EQ on all channels. The UB802 is well-suited for use with fixed audio and video installations as well as keyboards, samplers and computer application.

UB1202

The 10-input UB1002 features two mic preamps), while the 12-input 1202 is equipped with four. Both feature an effective, extremely musical 3-band EQ plus switchable low-cut filter on all mono channels. There is 1 post fader FX send per channel for external FX devices, main mix outputs plus separate control room, headphones and stereo tape outputs. Tape inputs are assignable to main mix or control room/phones outputs. These are extremely versatile mixers with possibilities ranging from connecting a DAT recorder and monitor speakers to running permanent video and audio installations.

FEATURES

- One (UB502), two (UB802, UB1002) or four (UB1202) XLR-balanced discrete inputs featuring studio-grade "Invisible Mic Preamps" with:
  - 130dB dynamic range for 24-bit, 192kHZ sampling rate inputs
  - Ultra-wide 60dB gain range
  - Lowest distortion 0.0007% (20Hz -20kHz)
  - A bandwidth ranging from below 10Hz to over 20kHz for crystal-clear reproduction of even the finest nuances
- Ultra-low noise ULN design, highest possible headroom, ultra-transparent audio
- Effective, extremely-musical 3-band EQ (2-band on the UB502) and peak LED on mono channel
- 5 (UB502), 6 (UB802) balanced high-headroom 1/4″ line inputs
- Main mix outputs plus separate headphones and stereo tape outputs
- Tape inputs assignable to main mix or phones outputs
- External power supply for noise-free audio and superior transient response

UB 802 Step-up Features:

- Switchable +48v phantom power for condenser mics
- 1 post fader FX send per channel for external FX devices
- 1 stereo aux return for FX applications or a as separate stereo input
- Main mix outputs plus separate control room, headphones and stereo tape outputs
- Tape inputs assignable to main mix or control room/phones outputs

UB1002/UB1202 Step-up Features:

- 10 (UB1002) or 12 (UB1202) balanced high-headroom 1/4″ line inputs with +4/-10 level selection on all stereo channels
- 3-band EQ plus switchable low-cut filter on all mono channels
- FX to control room function helps to monitor effects signal via headphones and control room outputs
- Long-wearing 60mm logarithmic-taper ALPS master fader and sealed rotary controls

UB1204-PRO • UB1204FX-PRO

12-Input 2/2-Bus Mic/Line Mixers

These are mixers whose small size belies their incredible versatility and audio performance. You get 8 balanced high-headroom line inputs, 4 Invisible Mic Preamps, 2 aux sends per channel (1 pre fader for monitoring applications and 1 post fader for external FX devices). There are 2 subgroups with separate outputs for added routing flexibility and 2 multi-functional stereo aux returns with flexible routing round off the list of high-quality features. The UB1204FX-PRO has all the same features plus a 24-bit digital stereo FX processor with 99 great-sounding VIRTUALIZER presets including reverb, delay, chorus, compressor, tube distortion, vinylizer and more, plus a 1kHz test tone generator. There are also solo and PFL functions on all channels as well as a pre/post fader switchable aux send for monitoring/FX applications.

STEP-UP FEATURES (FROM THE UB1202):

- Only 8 balanced high-headroom 1/4″ line inputs (not 12) with +4/-10 level selection on all stereo channels
- 2 aux sends per channel:
  1 pre fader for monitoring applications
  1 post fader for external FX devices
- Peak LEDs and mute/alt 3-4 function on all channels routes signal to subgroup instead of main outs
- Tape inputs assignable to main mix or control room/phones outputs
- 2 subgroups with separate outputs for added routing flexibility
- 2 multi-functional stereo aux returns with flexible routing
- Control room/phones outputs with multi-input source matrix
- Balanced main mix outputs with gold-plated XLR connectors plus separate control room, headphones and stereo tape outputs
- Internal switch-mode power supply for maximum flexibility (100-240v), noise-free audio, superior transient response plus lowest possible power consumption
- Includes rackmount brackets

UB1622FX-PRO

16-Input 2/2-Bus Mic/Line Mixer and 24-bit Multi-FX Processor

You get all the important elements right here in a 19″ form-factor. This compact mixer offers 12 balanced high-headroom line inputs, 4 Invisible Mic Preamps, 2 aux sends per channel (1 pre/post fader switchable for monitoring/FX applications and 1 post fader for internal FX or as external send). You also get an extremely musical 3-band EQ with semi-parametric mid-band plus switchable low-cut filter on all mono channels. The integrated 24-bit digital stereo FX processor offers 99 great-sounding VIRTUALIZER presets as well as a 1kHz test tone generator. There are 2 subgroups with separate outputs for added routing flexibility and 2 multi-functional stereo aux returns with flexible routing.

STEP-UP FEATURES (FROM THE UB1204 FX-PRO):

- Inserts on each mono channel for flexible connection of outboard equipment
- 3-band EQ with semi-parametric mid-band plus switchable low-cut filter on all mono channels
- 12 balanced high-headroom 1/4″ line inputs with +4/-10 level selection on stereo channels
- Peak LEDs, mute, main mix and subgroup routing switches
- Main mix outputs with jack and gold-plated XLR connectors, separate control room, headphones and stereo tape outputs

UB1832FX-PRO

18-Input 3/2-Bus Mic/Line Mixer and 24-bit Multi-FX Processor

If you are looking for a capable compact mixer with a breathtaking 3D XPQ stereo surround effect and a 9-band stereo graphic EQ, this is your machine. Of course, you also get 14 balanced high-headroom line inputs, 6 "Invisible" Mic Preamps, and 3 aux sends per channel and extremely musical 3-band EQs with semi-parametric mid bands plus switchable low-cut filters on all mono channels. There are 2 subgroups with separate outputs for added routing flexibility and 2 multi-functional stereo aux returns with flexible routing are also there for your convenience. You will also find the integrated 24-bit digital stereo FX processor with 99 great-sounding VIRTUALIZER presets.

STEP-UP FEATURES (FROM THE UB1622 FX-PRO):

- 6-XLR-balanced discrete inputs featuring studio-grade "Invisible Mic Preamps"
- 14 balanced high headroom 1/4″ line inputs with +4/-10 level selection on stereo channels
- 9-band stereo graphic EQ allows precise frequency correction of monitor or main mixes
- Breathtaking XPQ 3D stereo surround effect widens the stereo image and adds life and transparency to your sound—an easy way to put a unique final polish on your music and turn your performance into an incomparable experience.
- 3 aux sends per channel (Pre fader for monitoring, Pre/post fader switchable for monitoring/FX applications, Post fader for internal FX or as external send).
- Balanced TRS and gold-plated XLR main mix outputs, separate control room, headphones and stereo tape outputs

With their extensive and carefully thought-out routing possibilities, the FX-PRO mixers are perfect for both live and studio use. The prime suspects are high-quality recording studios as well as MIDI-studio applications, small-size PA applications (solo entertainers, duos to quartets, small bands), video edits and similar setups.

UB2222FX-PRO

22-Input 2/2-Bus Mic/Line Mixer and 24-bit Multi-FX Processor

The second-largest mixer in the UB Series, the UB2222FX-PRO has 16 balanced high-headroom line inputs, 8 Invisible Mic Preamps, 3 aux sends per channel, 3-band EQs with semi-parametric mid bands plus switchable low-cut filters on all mono channels, and integrated 24-bit digital stereo FX processor with 99 great-sounding VIRTUALIZER presets. There are 2 subgroups with separate outputs for added routing flexibility as well as 3 multi-functional stereo aux returns with flexible routing.

STEP-UP FEATURES (FROM THE UB1832 FX-PRO):

- 8-XLR-balanced discrete inputs featuring studio-grade "Invisible Mic Preamps"
- 16 balanced high headroom 1/4″ line inputs with +4/-10 level selection on stereo channels
- 3 multi-functional stereo aux returns with flexible routing

BEHRINGER

UB2442FX-PRO

24-Input 4-Bus Mic/Line Mixer and 24-bit Multi-FX Processor

This is the big cajuna of the whole UB Series. You get all the features you'd expect in an ultra high-quality compact mixer: 16 balanced high-headroom line inputs with dedicated gain controls on stereo channels 13-16, 10 studio-grade IMP "Invisible" Mic Preamps, and an effective, extremely musical 3-band EQ with semi-parametric mid band plus switchable low-cut filter on all mono channels. There is also an integrated 24-bit digital stereo FX processor with 99 great-sounding VIRTUALIZER presets.

STEP-UP FEATURES (FROM THE UB2222 FX-PRO):

- 10-XLR-balanced discrete inputs featuring studio-grade "Invisible Mic Preamps"
- 16 balanced high-headroom 1/4″ line inputs with dedicated gain controls on stereo channels 13-16
- 2 headphone outputs
- Channel inserts and direct outputs on each mono channel plus main mix inserts for flexible connection of outboard equipment
- 4 aux sends per channel: 2 pre/post fader switchable for monitoring/FX applications; 2 post fader (for internal FX or as external send
- 4 subgroups with separate outputs for added routing flexibility
- 4 multi-functional stereo aux returns with flexible routing
- BNC connector for standard gooseneck lamps

	UB502	UB802	UB1002	UB1202	UB1204 PRO	UB1204 FX-PRO	B1622 FX-PRO	UB1832 FX-PRO	UB2222 FX-PRO	UB2442 FX-PRO
Channels	1 mono+ 2 stereo	2 mono+ 2 stereo	2 mono+ 4 stereo	4 mono+ 4 stereo	4 mono+ 2 stereo	4 mono+ 2 stereo	4 mono+ 4 stereo	6 mono+ 4 stereo	8 mono+ 4 stereo	10 (8 + 2) mono + 4 stereo
Mono mic/line channel inputs	1 (XLR + 1/4″ TRS)	2 (XLR + 1/4″ TRS)	2 (XLR + 1/4″ TRS)	4 (XLR + 1/4″ TRS)	4 (XLR + 1/4″ TRS)	4 (XLR + 1/4″ TRS)	4 (XLR + 1/4″ TRS)	6 (XLR + 1/4″ TRS)	8 (XLR + 1/4″ TRS)	8 (XLR + 1/4″ TRS)
Stereo line channel inputs	2 (1/4″ TRS)	2 (1/4″ TRS)	4 (1/4″ TRS)	4 (1/4″ TRS)	2 (1/4″ TRS)	2 (1/4″ TRS)	4 (1/4″ TRS)	4 (1/4″ TRS)	4 (1/4″ TRS)	2 (1/4″ TRS & 2 (XLR + 1/ 4″ TRS)
Channel inserts (pre EQ, pre fader)	—	—	—	—	—	—	channel 1-4	channel 1-6	channel 1-8	channel 1-8
EQ mono channels	2-band	3-band	3-band + Low cut		3-band + Low cut		3-band mid-sweep + low cut			
EQ stereo channels	—	3-band	—	—	3-band	3-band	4-band	4-band	4-band	4-band
Phantom power (+48V)	—	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Main outputs	1/4″ TRS connectors		1/4″ TRS connectors		XLR connectors		XLR & 1/4″ TRS connectors			
Headphones output*	✓	✓	✓	✓	✓	✓	✓	✓	✓	2
2-Track I/O (RCA connectors)	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Control room outputs*	—	✓	✓	✓	✓	✓	✓	✓	✓	✓
PFL/solo section	—	—	—	—	—	✓	✓	✓	✓	✓
FX unit**	—	—	—	—	—	✓	✓	✓	with separate FX output	
Switch-mode PSU	—	—	—	—	✓	✓	✓	✓	✓	✓
Rack-mount kit inc.	—	—	—	—	✓	✓	✓	✓	✓	✓
Dimensions (W x D):	5¼ x 7″	7⅛ x 8⅔″	7⅛ x 8⅔″	9½ x 8⅔″	9 11/16 x 13 5/32″	9 11/16 x 13 5/32″	11⅞ x 13⅞″	16 1/16 x 14 1/16″	16 1/16 x 14 1/16″	16½ x 17¼″
Height (front/rear):	1½ x 1¾″	1½ x 1¾″	1½ x 1¾″	1½ x 1¾″	3 1/16 x 5″	3 1/16 x 5″	3 x 5″	3 x 5″	3 x 5″	2¾ x 5½″
Weight	1⅛ lbs.	2⅛ lbs.	2⅜ lbs.	3 lbs.	5⅝ lbs.	5¾ lbs.	7¼ lbs.	10⅜ lbs.	10⅝ lbs.	13 lbs.

** 1/4″ TRS connector ** with footswitch connector*

DDX3216

Fully Automated 32-Channel, 16-Bus Digital Mixing Console

The DDX3216 offers a feature set you won't find in any other product in its class. 32 channels, 16 internal busses and 8 aux sends. Each channel features a fully parametric 4-band equalizer, sweepable high-pass filter, gate, compressor/ limiter and phase inversion—all simultaneously operable. The first 16 channels also offer a delay function. Add four on-board effects processors that put just about any imaginable effect type at your fingertips, in top quality — and simultaneously accessible. Then full-fledged dynamic and static automation functions, extensive synchronization options and MIDI implementation, a comprehensive internal I/O patchbay, full digital compatibility and functionality. Plus unusually flexible connectivity and expandability, supported via two expansion slots for ADAT, TDIF, AES/EBU interfaces and more. All based on 32-bit floating-point processing (40-bit for the EQ's).

This all comes in a rack-mountable case and with an intuitive, logically laid out user interface based on excellent software and effective control elements, like motorized ALPS faders, assignable channel controls with LED rings, push-and-turn master controllers and a large, backlit LCD display. Ideal for almost any application including recording, live sound, multimedia, audio for video, broadcasting and installations.

FEATURES

- 32 full-fledged channels and 12 high-end ultra low-noise "Invisible" Mic Preamps with phantom power deliver crystal-clear audio with incredible reproduction of the slightest sound nuances and ample headroom.
- 16 busses, 8 aux sends and internal input/output patchbay for comprehensive routing options
- Ultra high-resolution 24-bit AKM A/D and CRYSTAL D/A converters provide optimum audio quality bound to please even the most discerning user.
- High-power floating point DSP technology (32-/40-bit) ensures virtually unlimited internal dynamic range
- Fully featured dynamic and snapshot automation
- 4-band fully parametric EQ, compressor, gate, sweepable high-pass filter and phase inverter on all 32 channels — with additional delay function on the first 16 channels — provide effective dynamic processing.
- Four (simultaneously operable) ground-breaking effects processors, accessible from all 32 channels, provide first-class algorithms, such as chorus, delay flanger, LFO filter, phaser, pitch shifter, reverb, tremolo, and many more.
- 17 precise, low-noise 100mm motorized ALPS faders
- Additional compressor/limiter (switchable pre/post) and EQ for stereo main mix
- Freely configurable built-in level meters on all channels and channel controls with LED rings facilitate editing of any of nine selectable parameters per channel.
- Four freely assignable analog outputs on balanced 1/4" TRS and MIDI plus RS232 connectors round out the I/O options by allowing interfacing with a wide variety of external devices.
- Extensive MIDI implementation capability enables program and control changes, MIDI sysex and MMC.
- Six master controllers with comfortable push-and-turn functionality
- Analog feel, intuitive user interface
- Large, easy-to-read LCD display with adjustable contrast
- Synchronization via SMPTE, MTC or internal clock
- Dither, word length and noise shaping adjustable for digital main outputs
- PCMCIA slot for saving/loading various libraries and other settings
- Routing of complex signal configurations is accomplished by an internal input/output patchbay.
- Two extension slots for installation of optionally available digital interfaces for virtually unlimited connectivity:
  - ADT1616 16-channel ADAT interface
  - TDF1616 16-channel TDIF interface
  - AES808 8-channel AES/EBU interface

MACKIE

1202-VLZ PRO • 1402-VLZ PRO

12 x 2 and 14 x 2 Compact Mixers

Providing maximum performance in minimum sizes, the 1202-VLZ Pro and 1402-VLZ Pro are the mixers of choice for tens of thousands of musicians and sound engineers who need equipment to serve "double duty" in studios and on the road. With 12 or 14 balanced input channels divided between 4 (1202) or 6 (1402) mono mic/line inputs and 4 stereo line input channels, these mixers excel in the field or in the studio. Video suites doing dialog recording, voiceover or sound effects and project studios will find the range of bussing flexibility a blessing. For live gigging, the rugged steel chassis will stand up to the abuse of the road and the 2 aux sends are perfect for monitor and or effects sends.

FEATURES

The1202 and 1402-VLZ PRO are the standard for small-to-medium video post production suites. They're the most popular mixers for home project studios. They're in touring racks of performers. They record location sound for major Hollywood movies. They're on duty at major-league football games, recording the crunch of bodies and the sportscasters' breathless play-by-play. They are also used as live field mixers by the news departments of every major network.

Input Channels

- 4 mono balanced XLR mic/ 1/4″ line inputs with inserts, trim and rotary level controls.
- 4 balanced 1/4″ stereo line inputs with rotary level controls.
- 2 Aux sends per channel can be used as effects sends or for creating monitor mixes and offer 15dB of extra gain above Unity
- Each input has 3 bands of EQ at 80Hz, 2.5kHz and 12kHz with a ±15dB boost/cut.
- Inputs have a -18dB/oct. 75Hz Lo Cut filter that eliminates stage rumble, wind noise, P-pops and other low frequency noise
- The Mute/Alt 3-4 buttons located on each input channel serve two functions: muting the input channel from the main mix, and signal routing, where they act as your gateway to an extra stereo bus
- Constant-loudness pan pots on each input channel keep sound at a consistent volume when panning between left, center and right positions
- PFL Stereo in-place Solo on each channel

Aux Inputs

- 2 balanced 1/4″ stereo returns with EFX to Monitor switch
- Aux 1 master control with Monitor/Post assign switch
- Stereo RCA 2 track tape return

Outputs

- Main stereo outputs are XLR-balanced switchable from +4dBu to mic level
- 1/4″ balanced alternate outputs (alt 3/4) are assignable from mute buttons on input channels
- 2 balanced 1/4″ auxiliary sends
- 1/4″ TRS balanced stereo control room outputs
- Stereo RCA (2 track) tape outputs
- Multi-input source matrix with level control lets you route any combination of Main Mix, Tape In and Alt 3-4 to the Control Room/Phones bus. Routings can be used for submixes, monitoring or tracking and can be subsequently routed to the main mix

Additional Features

- Gain knobs have Unity gain detents, when input levels are properly set, the highest headroom and lowest noise floor is at unity gain
- 12 segment LED's give accurate level status of main outputs or soloed channels
- 48-volt phantom power for condenser mics
- Built-in power supply–no wall wart!
- Rude Solo Light LED bluntly advises that a channel is soloed
- Tape input level control and Tape To Main Mix switch
- 3-year parts and labor warranty.
- Optional RM1202VLZ rackmount ears

1402-VLZ PRO *Step-up Features*

- 6 mono balanced XLR mic/1/4″ line inputs instead of 4
- Switchable AFL/PFL Solo
- Optional RM1402VLZ rackmount ears
- 60mm long-wearing log-taper faders ensure a consistent, smooth accurate response through the length of the fader travel

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

MACKIE MIXER TECHNOLOGY

Mackie revolutionized the compact mixer market by applying features and audio quality previously reserved for high-end, large format mixers. From the smallest to the largest of their models, Mackie puts their best effort into every phase of board design. For example, XDR (Extended Dynamic Range) mic preamps offer over 130dB of headroom, exceeding the performance of outboard preamps costing $1000 to $2000 per channel. Project studios, video suites, multimedia authors, live venues, broadcast facilities and houses of worship seeking uncompromising sonic quality and maximum flexibility will find what they are looking for in Mackie's VLZ PRO mixer line.

Very Low Impedance

- Very low impedance is achieved by scaling down resistor values by a factor of three or four, resulting in a corresponding reduction in thermal noise.

XDR Mic Preamps

- 130dB dynamic range and extremely flat frequency response allows the XDR mic preamps to handle inputs from 192kHz workstations without added coloration.
- DC pulse transformer reject RFI without attenuating frequencies of 15kHz and above.
- Ultra-high-speed, large-geometry input diodes protect the XDR mic preamps from hot-patching and direct short circuits in cables carrying phantom power.

EQ

- Positioned at 12kHz, the Hi Shelving EQ adds sheen and presence to instruments and vocals, enhancing the textures of sounds without contributing to aural fatigue.
- Positioned at 80Hz, the Low Shelving EQ emphasizes the fundamental lows of kick drums, and other bass instruments while allowing you to fatten up male vocals and instruments like guitars.
- Mid-range frequency is placed at 2.5kHz (1202/1402-VLZ PRO), harmonics of vocals and instruments are enhanced without becoming strident or fatiguing.
- Sweepable Mid-Band EQ (1604/1642-VLZ PRO) from 100Hz to 8kHz for specific EQ treatment of a broad range of frequencies.

Mix Amp Headroom

- Negative gain mix amplifier architecture sets standard mixing levels at -6dB. This offers a greater amount of headroom, allowing up to 4 times as many hot signals to be summed at the main mix bus without clipping

Impact Resistance

- Knobs ride just thousandths of an inch from the metal surface of a mixer chassis.
- Brutal knob impact is absorbed by broad pressure on a tough circuit board
- Ultra-tight lip seal design provides a continuously-sealed barrier against dust and liquid
- A braced, horizontal circuit board and shock-absorbing structure eliminates force transferred to the circuit board.

1642-VLZ PRO *16 x 4 x 2 Compact Mixer*

Bridging the gap between the 1604-VLZ Pro (next page) and the smaller 1402-VLZ Pro, the 1642-VLZ Pro gives you 4-bus mixing flexibility, ten XDR mic preamps and four stereo line-level channels—fewer mic preamps and more line-level inputs at your fingertips. Yet it also has live-sound features like EFX to monitor, AFL/PFL solo, pre-fader aux sends, and built like a tank construction. The ideal mixer if you record with lots of keyboards, samplers and drum machines. Perfect for home and project studios as well as video post-production applications.

All The Features of the 1402 VLZ Pro ***PLUS—***

Input Channels

- 16 input channels total— channels 1-8 are mono and channels 9 - 16 are stereo.
- Each channel features input trim, 4 aux sends (aux 1 & 2 assignable pre/post), -20dB pad, 75Hz HPF, Solo (PFL and Solo-In-Place), mute, pan, bus 1/2, 3/4 and L/R assignment buttons and overload LEDs
- Channels 1 - 8 feature balanced XLR mic/ 1/4″ line with inserts, direct outputs and 3 band EQ
- 2 stereo balanced XLR mic/ 1/4″ line inputs, 2 balanced stereo 1/4″ line inputs
- 4-band active EQ on stereo channels

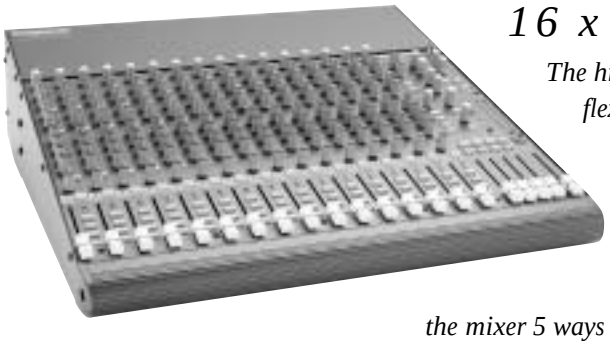
Outputs

- Four 1/4″ balanced subgroup outputs have 60mm faders, switchable L/R assignment to the main mix bus.
- Two 1/4″ stereo headphone outputs
- Four 1/4″ balanced aux sends. Aux send 1-2 master section with level controls and solo
- 1/4″ balanced mono output with level control

Additional Features

- Four 1/4″ balanced stereo aux returns with level controls and solo.
- Switchable AFL/PFL Solo with level control
- Optional rack ears are allow mounting in a standard 19″ rack.

1604-VLZ PRO



16 x 4 Compact Mixer

The highest headroom and lowest noise of any mixer in its class, the 1604's flexible 4-bus architecture is ready for just about any project recording studio or live venue. Post production and broadcast facilities will also appreciate the myriad routing possibilities. VLZ design ensures quiet mixes with all 16 channels open during mix down, while the XDR mic preamps reveal all of the subtleties and nuances of your favorite condenser mic. Mackie's rotopod design and supplied rackmount hardware allows you to configure the mixer 5 ways by allowing you to rotate the input and output jacks to the top, back or front of the mixer with or without rackmounting. With unprecedented sonic performance, bandwidth, dynamic range and versatility, the 1604-VLZ meets the demands of the most sophisticated digital audio workstations —while remaining totally affordable.

ALL THE FEATURES OF THE 1642 VLZ Pro *PLUS*—

Input Channels

- 16 mono balanced XLR mic/ 1/4″ line inputs with inserts, input trim, 6 aux sends/4 per channel (aux 1 & 2 assignable pre/post), -20dB pad, Solo (PFL and Solo-In-Place), mute, pan, bus 1/2, 3/4 and L/R assignment buttons and overload LEDs
- Each input channel features 3-band active EQ with sweepable midrange (12kHz high shelving, mid eq sweep between 100Hz - 8kHz, 80Hz low shelving eq ±15dB) and 75Hz HPF
- Input channels 1-8 feature 1/4″ balanced direct outputs

Outputs

- 1/4″ balanced main outputs with inserts
- 6 balanced 1/4″ auxiliary send
- Single 1/4″ headphone output with level control

Additional Features

- 5 physical configurations via rotating I/O pod
- Rack mount kit included

DFX-6 / DFX-12

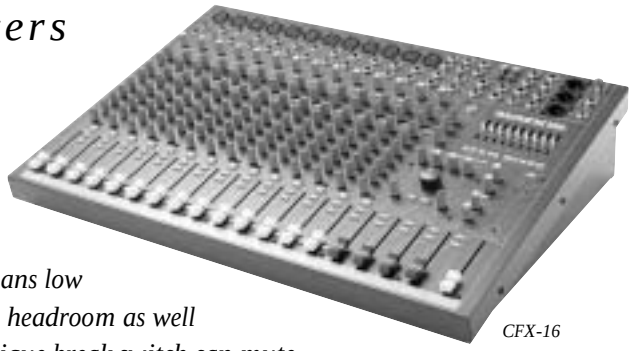
Compact Live Sound Mixers with EFX

Combining Mackie's sound quality with an extensive feature set, the DFX-6 and DFX-12 are designed specifically for solo or small performing groups that don't have the luxury of a front-of-house sound technician. All knobs are color-coded and can be easily identified at a glance. Channel faders are white. Effects and Monitor Sends are red. Master level faders are blue. The jack field is on the top surface so connections are immediate and every channel has a level set and overload LED that can be easily seen on a dark stage. Both include a combination of mic/line and stereo line inputs, silky smooth 60mm faders, and Mackie's proprietary 32-bit EMAC custom digital effects processor with 16 effects. Each also offers low-noise, low-distortion, studio-grade mic preamps wrapped in a box rugged enough to withstand years of on-the-road wear and tear.

- Low noise, high headroom mic preamps with switchable phantom power
- 5-band stereo graphic equalizer with 12dB boost/cut, Bypass and Aux/Main Assign
- 2-band channel strip shelving EQ at 80Hz and 12kHz
- 75Hz Low Cut filters and inserts on mono mic/line channels
- 32-bit Digital EFX Processor with 16 effects
- Input trim controls with LED level set indicators
- Separate Aux Send and Effects Send for each channel
- 2 stereo Aux Returns
- Master Aux 1, Aux 2/EFX and CD/Tape Return faders
- Effects to Monitor feature with separate rotary level control
- Tape/CD inputs and Tape outputs
- Vocal Eliminator switch for "Karaoke" use
- Break switch for playing music between sets
- Headphone output with level control
- Bal./unbal. XLR and 1/4″ main stereo outputs

CFX-SERIES

12-, 16- and 20-Channel Audio Mixers with Digital Effects



CFX-16

Available in 12 (CFX-12), 16 (CFX-16) and 20-channel (CFX-20) configurations, the CFX mixers incorporate digital effects and a 9-band graphic EQ to offer all-in-one compact mixing solutions. Sounding clean and realistic, the effects processor rivals many outboard processors, while the graphic EQ on the output section is of "audiophile-quality" which means low noise, free of phase distortion. The mixers also feature low noise and high headroom as well as balanced I/Os and multiple busing options. For live performance, a unique break switch can mute the entire mixer (except for the two-track return input) as well as the effects and graphic EQ sections.

Input Channels

- Level-setting LED for each channel.
- Low-noise, high-headroom mic preamps with 50dB of gain; 48v phantom power
- +30dB of line level gain and a full 15dB of attenuation to "pad" hot signals.
- Two Aux Sends with balanced 1/4″ outs, switchable to pre-fader for monitor use and post-fader for use as effects sends
- One external and one internal EFX Send.
- Pan, Mute, PFL solo on each channel
- Pre-Fader Solo switch allows channel monitoring via headphones
- 3-band EQ (12kHz shelving HF, ultra-wide 100Hz to 8kHz bandwidth sweepable peaking Midrange EQ and 80Hz LF).
- Inserts and 100Hz 18 db/octave low-cut filters on all mic/line channels
- 4-band EQ on stereo line channels

Subgroup

- Four Subgroup buses with 1/4″ balanced direct outputs and L/R assign "collect" channel signals assigned to them so you can submix vocals, drums and other audio signals for monitoring or tracking.

Effects

- 32-bit digital FX with 9 reverbs, 4 delays, phaser, chorus, flange; 2 parameter controls and an EFX wide spatial expander and bypass switch

Master Section

- Studio-quality, 9-band stereo graphic EQ
- Tape/CD inputs with level control assignable to Main Mix via Break switch
- RCA tape/CD inputs and tape outputs.
- Two stereo effects returns with balanced 1/4″ inputs

Outputs

- Balanced/unbalanced 1/4″ TRS and balanced XLR outputs for mains with inserts.
- Headphone jack w/level control.
- XLR balanced subwoofer output from built-in 18dB/oct. 75Hz crossover!
- Extra 1/4″ balanced utility stereo outputs with level control (post-Main Fader).

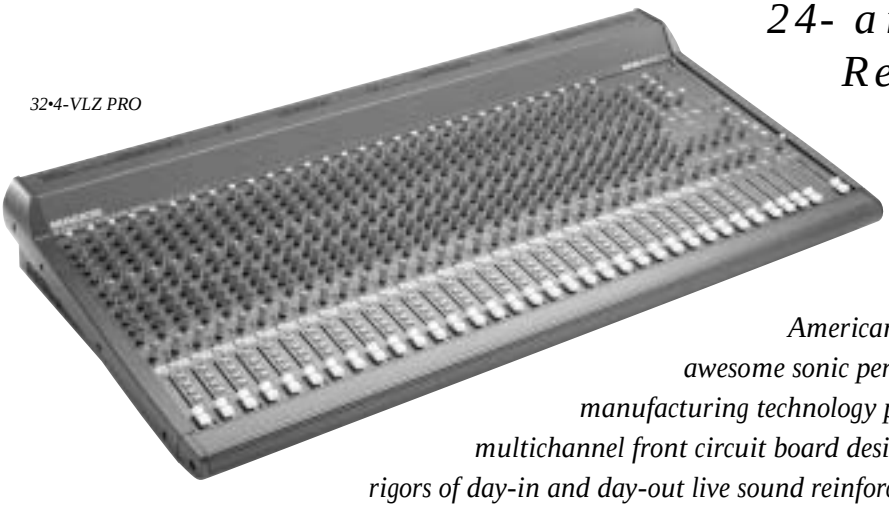
Conveniences

- Break Switch mutes all channels and routes Tape input to the mains while you're on break so you don't have to worry about feedback during breaks
- Long-wearing 60mm logarithmic taper faders give you smooth, linear control throughout the fader's entire travel.
- Rude Solo Light alerts of any current soloed channels

	1202-VLZ Pro	1402-VLZ Pro	1642-VLZ Pro	1604-VLZ Pro	CFX-12	CFX-16	CFX-20
Total Channels	12	14	16	16	12	16	20
Buses	2	2	4	4	4	4	4
Mono Inputs	4 Mic/Line	6 Mic/Line	8 Mic/Line	16 Mic/Line	8 Mic/Line	12 Mic/Line	16 Mic/Line
Stereo Inputs	4 Line	4 Line	2 Mic/Line, 2 Line	–	2 Line	2 Line	2 Line
XDR Mic Preamps	✓	✓	✓	✓	–	–	–
EQ Mono Channels	3-Band (Fixed)	3-Band (Fixed)	3-Band (Mid-Sweep)	3-Band (Mid-Sweep)	3-Band (Mid-Sweep)	3-Band (Mid-Sweep)	3-Band (Mid-Sweep)
EQ Stereo Channels	3-Band (Fixed)	3-Band (Fixed)	4-Band (Fixed)	–	4-Band (Fixed)	4-Band (Fixed)	4-Band (Fixed)
Aux Sends/Channel	2 Mono	2 Mono	4 Mono	6 Mono	2 Mono	2 Mono	2 Mono
Aux Returns	2 Stereo	2 Stereo	4 Stereo	4 Stereo	2 Stereo	2 Stereo	2 Stereo
Direct Outputs	–	–	8	8	–	–	–
Main Outputs	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.
Mono Output			1/4″ Bal.	1/4″ Bal.	XLR Bal.	XLR Bal.	XLR Bal.
Internal Effects	–	–	–	–	✓	✓	✓
9-Band Graphic EQ	–	–	–	–	✓	✓	✓

MACKIE

24•4-VLZ PRO/32•4-VLZ PRO



32•4-VLZ PRO

24- and 32-Channel Sound Reinforcement Consoles

First introduced almost five years ago (but updated many times since), Mackie's SR24•4 and SR32•4 have totally redefined the 24-and 32-input/4-bus mixer. Feature-packed, extremely affordable, American-made mixing consoles, complete with awesome sonic performance, they incorporate the very latest in manufacturing technology plus Mackie's own proven horizontal, multichannel front circuit board design. They are also designed to stand up to the rigors of day-in and day-out live sound reinforcement, yet still have the required dependability and pristine audio quality for professional studio applications as well.

The SR24•4 and SR32•4 provide superior mixing and recording. They feature major headroom, thanks to Mackie's VLZ circuitry. Main and submix amplifiers use distinctive negative-gain mix architecture to double the mix headroom of conventional designs. Ultra-quiet high-headroom mic preamps on 20 (SR24•4) or 28 (SR32•4) mono channels mean great-sounding vocals or instruments. Low-cut filter (on mono channels) centered at 75Hz allows you to cut out mic thumps and wind noise. Four buses make live mixing a breeze by letting you control the overall level of those channels with one fader. Six aux sends per channel offer the flexibility to handle multiple chores at once. Two aux sends are switchable to pre- or post-fader so you can have 4 pre-fader aux sends, or 4 post-fader aux sends. They also feature a sweepable midrange EQ (on mono channels) with midrange centering anywhere from 100 Hz to 8 kHz.

FEATURES

- They combine premium XDR mic preamps with 60dB gain range, 130dB dynamic range for hot 24-bit digital signals, massive +22dB input handling capacity, 192kHz bandwidth and 0.0007% THD with the best RFI resistance of any compact SR mixer in the galaxy thanks to advanced DC-pulse transformers.
- 20 (SR24) and 28 (SR32) XDR mic preamps are impedance-independent, so the frequency response remains constant whether the mic preamp is presented with an extremely high- or low-impedance load. Additionally each channel has its own switchable low-cut filter (18dB/octave at 75Hz), allowing you to cut out mic thumps and wind noise that can enter your mix and rob you of amp power.
- They have 20 (SR24) and 28 (SR32) mono mic/line input channels, with XLR mic inputs and 1/4″ TRS line inputs, and 2 stereo line input channels with 1/4″ TRS inputs.
- Each channel strip has an input trim, 6 aux sends (2 prefader, 2 post-fader and 2 switchable pre or post), 3-band EQ with sweepable mid-range (4-band EQ on the stereo channels), pan control, and mute, solo and bus assign switches. A 60mm fader provides output gain for each channel.
- The mono channel EQs provide a range of ±15dB at the following frequencies: 12kHz shelving high-frequency EQ, 100Hz to8kHz sweepable peaking mid-frequency EQ, and 80Hz shelving low-frequency EQ.
- Stereo channels (21-24) provide ±15dB of boost and cut at the following frequencies: 12kHz shelving high-frequency EQ, 3 kHz peaking high-mid frequency EQ, 800Hz peaking low-mid frequency EQ, and 80Hz shelving low-frequency EQ.
- You can mix any combination of channel strips down to a single submix bus—or any combination of the four submix buses. Channels can be assigned to buses 1-2, 3-4, and Main Mix L/R, and the 4 sub can be assigned to Left and Right Main Mix, or fed directly to a multitrack recorder. This is incredibly helpful, not only in a live situation, but in the studio as well. Furthermore, each bus is "double-bused" providing 8 eight outputs that can be connected to an 8-track recorder without repatching anything.
- Each sub out (1-4) and main out (L/R) has a TRS insert jack. Furthermore, Mackie's unique "AIR" EQ circuit is included on each of the four subgroups. This peaking EQ circuit, centered at 16kHz, enhances guitars, vocals, percussion, whatever, giving a gentle "lift" to the extreme high end. The result is extra detail and fidelity in the high end.

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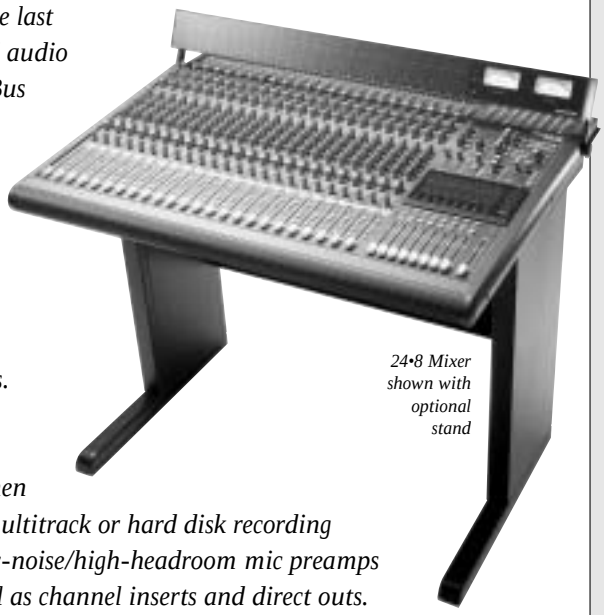
EQUIPMENT LEASING AVAILABLE

24•4-VLZ PRO/32•4-VLZ PRO

- Outputs include XLR and 1/4″ TRS line outputs for the left and right mains, 1/4″ TRS line outputs for subs 1-4, and an XLR mono main output. The mono main out has its own level control so a mono mix can sent to another zone and adjusted accordingly.
- The Phones/Control Room switch and level control are connected to stereo headphone outputs and left and right Control Room output, allowing the stereo Tape Return, Left/Right Main Mix, and Solo to be monitored.
- A stereo playback device can be monitored via the Tape Return inputs. Tape Return to Phones/C-R routes the tape playback signal into the monitor system and meters, and the Tape Return knob adjusts the level of playback, which can be monitored via headphones.
- Tape return signals can also be assigned directly to the Main Mix. This not only routes Tape Returns to Main L/R outputs but also disables all other inputs to the mains. You can play a tape or CD between sets without losing a channel and submaster settings. They also have RCA-style tape outs for output to conventional stereo recording devices.
- Each of the six aux sends has its own individual master send control, driving 1/4″ TRS output jacks. Six stereo aux returns are provided, with 1/4″ TRS input jacks. Two aux returns can be folded back into AuxSends 1 and 2 via their own volume controls to add effects in stage monitors.
- An XLR input is provided for a talkback mic, which can be assigned to the Main Mix (for making announcements over the mains) or Aux 1-2. The talkback mic has a level control in the talkback section.
- Like all Mackie mixers, the SR24•4 and SR 32•4 are designed for rugged 24-hour-a-day use. With their multiple input/output configurations, true 4-bus architecture, 6 aux send and extensive routing capabilities, they can be used in a variety of live sound and recording applications. Their sturdy steel construction houses rugged, double-sided SMT-plated fiberglass circuit boards, and 60mm faders with ultra-tight lip seals for keeping out dust and other contaminants. Impact-resistant knob are mounted so they "ride" just above the steel chassis.

16/24/32•8

8-Bus Recording/PA Mixers



24•8 Mixer shown with optional stand

Designed to eliminate the last barrier between you and audio creativity – Mackie's 8-Bus mixers have recorded more platinum albums and major motion picture soundtracks than any other mixer in their class – in fact, they set the standard for affordable 8-bus consoles. Excellent for project studios and digital multitrack recording, when combined with digital multitrack or hard disk recording systems, they feature low-noise/high-headroom mic preamps on every channel, as well as channel inserts and direct outs. In the channel strip you'll find Mackie's truly musical EQ – with Hi and Lo shelving EQ, parametric Hi-Mid, and sweepable Low-Mid EQ – as well as a Low Cut Filter, which allows you to eliminate mic thumps and room rumble from your mix, and use your EQ for music and vocals. With Mix B, "in-line monitoring" is a breeze. Six aux sends and six aux returns allow you to use all the effects you want. And the 8•Bus's complete talkback and phones level controls make communicating with the talent a snap. And besides being built like tanks, the 24•8 and 32•8 can grow along with your budget and input requirements. Both can be expanded in increments of 24 channels with the 24E Channel Expander Console. You can also add meter bridges, stands, and even a "sidecar" for patchbays.

	32• Bus	24• Bus	16• Bus
Total Channels	32	24	16
Buses	8	8	8
Mix B Inputs	32	24	16
Mic Preamps	32	24	16
EQ	4-band parametric	4-band parametric	4-band parametric
Aux sends/ch.	6	6	6
Aux Returns	4	4	4
Tape Outputs	24	24	24
Channel Inserts	32	24	16
Faders	100mm	100mm	100mm

MACKIE

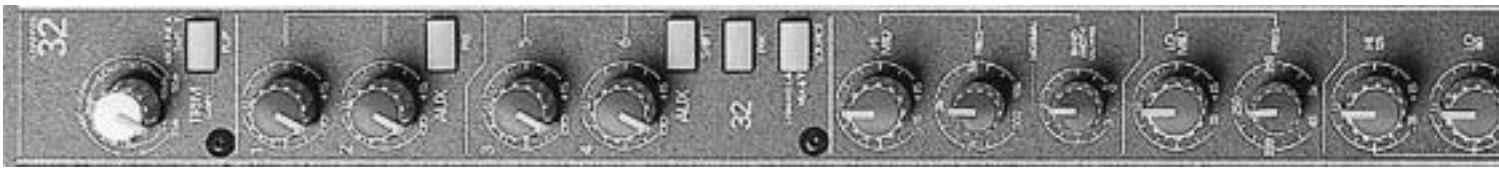
16•8 / 24•8 / 32•8 8•BUS MIXERS

Channel Strip

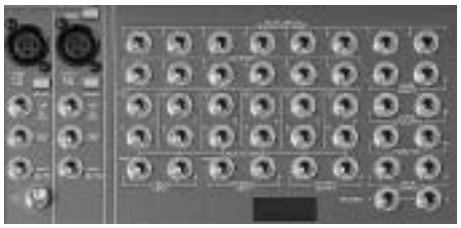
- In-line FLIP reverses tape and mic/line inputs between channel strip and Mix-B/Monitor section
- Each channel strip has 6 mono auxiliary sends with several routing options:
- Auxes 1 & 2 have two independent level controls that share a common Pre/Post switch.
- Aux 3-4/5-6 are accessible from two level controls: A 'shift' button changes Auxes 3 4 to 5-6. A 'source' switch selects signal source of Aux 3-4/5-6 from channel strip or channel's Mix B/Monitor send so you can build and effects mixer (pre or post-Mix-B level) to assign to phones during tracking.
- True parametric, 3-control Hi Mid EQ with ±15dB boost/cut: Ultra-wide 500Hz to 18kHz frequency sweep range; bandwidth can be adjusted from a very wide 3-octave width to a very narrow 1/12-octave width.
- Low Mid EQ with ultra -wide 45Hz to 3kHz sweep, ±15dB boost/cut
- ±15dB shelving Hi (12kHz) and low (80Hz) EQ
- Multi-purpose 18dB/oct. low cut filter cleans up "mix mud", cuts PA rumble, creates a "neo-peaking" bass control when used with LO shelving boosts
- Independent Mix-B (Monitor) section—think of it as as discrete "channel strip within a channel strip"—with its own pan, level, EQ and source capabilities. During mixdown, use as extra pre-fader stereo Aux send or double your inputs
- Mix-B Split EQ assigns Hi and LO EQ to Mix-B
- Mix-B Source selects from flip switch or channel strip (pre-fader) for in-line monitoring while recording or gives an extra stereo aux bus
- Constant power, buffered panpot for rock-solid panning
- +22dB Overload LED (monitors three critical points in the signal chain, and displays the one that's highest at any given time) and Hyperactive -20dB Signal Present LED (can also tell you at a glance what vocal or instrument is on what channel)
- Selectable Solo with Channel Metering allows soloing in full stereo perspective: displays soloed channel operating level on master L/R meters so input trims can be adjusted for optimum levels.

Top of the Channel Strip (not shown)

- DC phantom power for condenser microphones is applied to channel strips in groups of eight. It has a "ramping" function that gradually increases from 0 to 48v power when you turn it on preventing thumps and pops.
- Two BNC sockets for 12v gooseneck lamps
- Balanced mic inputs lead to the VLZ mic preamps. Mackie's VLZ (Very Low Impedance) design provides terrific dynamic range with an EIN of -129 dB and 10Hz to 300 kHz bandwidth.
- Balanced/unbalanced 1/4″ jacks with 40dB range handle everything from hot digital inputs to older keyboards with low output voltages.
- 1/4″ 4dB direct out jack (connected to the output of the channel, post-EQ/post-fader/post-mute switch
- Channel Insert allows you to insert external serial processing equipment (eg. compressor, gate) into the main signal path of the input.



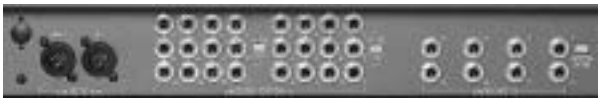
Master I/O Section



- Main and submaster inserts allow you to insert a compressor or EQ into main L/R or any of the 8•Bus submaster circuits.
- Aux Returns are unbalanced stereo with a L&R input for each return channel. If you only have one return signal, plugging it into the Mono/L jack will cause it to be connected t both the left and right return inputs and end up centered in your stereo image.
- Control Room output, Main Mix output, Mix-B output and Studio output
- 2-Tk Input and External Inputs offer you the ability to listen to two stereo sources directly, without patching through input channels.

Rear Panel

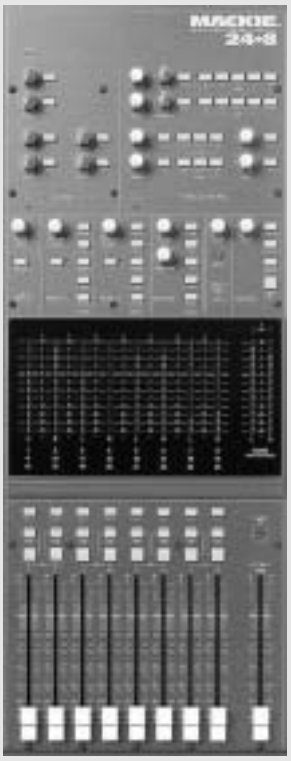
- Three tape output jacks per bus (total of 24). Balanced outputs, switchable from +4/-10.
- +4dBu balanced Tape Returns, switchable to -10dBV in banks of 8 returns
- "Triple busing" allows you to feed a 24-track deck without having to constantly repatch. When you send a signal to Submaster 1 output, for instance, it will appear as Submaster Outputs 1, 9, and 17. If you have a 16-or 24-track deck, simply patch these outputs to the corresponding multitrack inputs. Now the tracks on your multitrack that are in Record mode will accept the signal, while the tracks in Safe mode won't.



16•8 / 24•8 / 32•8 8•BUS MIXERS

Master Section

- Six Stereo Aux Returns. All have 20dB gain, Solo, and can be used in stereo and mono. Aux return 1 & 2 are pannable and bussable. 3 & 4 are assignable to the phones for 'wet' monitoring.
- 6 Aux Sends with Solo and Solo LED
- Talkback assigns to all submasters, main mix, Aux 1, Aux 2 or Phones 1&2
- Solo level adjust and ultra-rude LED
- Monitor section with separate control room and studio levels, source selection between L/R mix, Mix-B, tape and external. Can be switched to mono.
- Two separate headphone sections can be used totally independent of each other. Each features source selection between control room and any combination of Aux 3/4, Aux 5/6, Mix-B or external source. Solo allows control room to hear what musicians are hearing in their headphones.
- Mix-B/Monitor section can be used as an independent stereo out for PA monitor mix, 2-track recording, video/broadcast feed or assigned to L/R mix.
- -40 to +10 bar graph LED displays for each sub-master and Solo/Main (with main L/R +22dB Clip LEDs)
- Expansion consoles let you add channels in banks of 24 to either the 24•8or 32•8. Expanders have their own internal mix amps so the main board only "sees" one extra channel per expansion console.
- Trick Bus Solo switches send odd-numbered buses to the left speaker and even-numbered buses to the right speaker—unless you've pressed the respective Mono L/R button. When a bus has been mono-ed, Solo sends the bus to both speakers
- L Mix/R Mix and Mono L&R buttons assign buses to main L/R stereo bus
- Built-in Talkback Mic with level control lets you scream engineering commands to any combination of Aux Send 1, Aux Send 2, Tape/Submasters (L/R Mix and the eight submasters) and Phones/Studio. When Talkback is engaged, control room outputs are padded by 20dB to avoid feedback.



8•BUS SERIES ACCESSORIES

Meter Bridges

Matching meter bridges (MB•16, MB•24, MB•32) are available for each mixer. Each provides 12-segment LED ladders for each input channel, and lighted VU meters for master Left/Right output. Like the mixers' integral meter sections, the LEDs in the bridges are with Lexan to resist dust and humidity. On each meter bridge, input buttons allow you to globally switch between Tape Return input and Channel Strip post-fader output. Each meter attaches in minutes and can be tilted as desired (up to 90°). They also tilt down fully flat to save you inches of road case depth—and from having to buy another case just for the meter bridge.

The Stand

Any 8•Bus (including the 24•E Expander) can be floor-mounted via the optional The Stand. Made of rugged steel with ultra-tough, 2-part polyurethane coating, The Stand assembles in minutes.

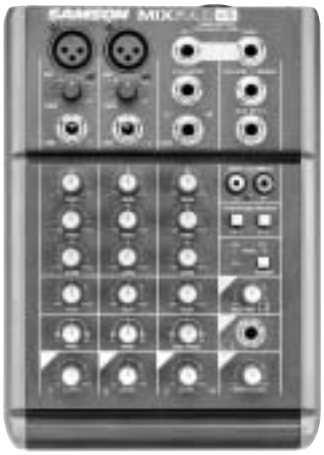
24•E Expander Console (for 24•8, 32•8 only)

The 24•E is essentially another 24-channel 8•Bus mixer—but without a master output section. You get 24 complete input channel strips and 24 more tape returns. The 24•E connects to the 24•8 or 32•8 via a multi-pin connector. Additional 24•E's can be daisy-chained to create 48, 56, 72, 80 or 96 more input channels. The 24•E's outputs are pre-mixed in the expander to reduce line and thermal noise and to maintain maximum sonic quality at the main console. The connecting cables are long enough so that you can place one or Expanders on each side of your main 8•Bus mixer.

SAMSON

MIXPAD 4

Compact 4-Channel Audio Mixer



An incredibly compact 4-input stereo mic/line mixer with exceptional sound and a host of features, the Mixpad 4 also offers AC/DC capability, making it ideal for use in ENG and field recording applications as well as in the studio. Exceptionally professional from its high headroom design (+27dB) to its durable extruded aluminum chassis, the Mixpad 4 also shines as a sub-mixer in live and recording settings.
Features include 3-band EQ and pan/balance and level controls on each channel, a front-panel stereo headphone jack, two balanced mic/line inputs (XLR and 1/4˝ connectors), 1/4˝ stereo line input, CD/tape input, 48-volt phantom power, an Aux send and a stereo Aux return (with level control which can be used as 2 mono aux returns), as well as a main level control and power and peak LEDs.

- Two balanced (XLR and 1/4˝) mic/ line and one stereo input (4 channels total)
- Trim controls on mic input channels
- 48v phantom powered mic inputs
- Independent 3-band EQ for each channel, with ±15 dB of cut/boost for low (80 Hz) and high (12 kHz) frequencies, and ±12 dB of cut/boost for mid (2.5 kHz) frequency
- Independent CD/tape input; signal is routed via channel EQ
- Constant-level pan controls (mono channels) and balance controls (stereo channel)
- One Aux send (pre-fader)
- One stereo effects return with front panel level control
- Peak overload LEDs on left and right main outputs for monitoring
- High slew rate allows it to react quickly to transients and to maintain crisp sound.
- Balanced stereo output with 1/4˝ jacks for connection to amps and other gear
- Front-panel headphone output
- An inline power supply maintains the mixer's low profile while eliminating hum.
- Battery operated, the mixer can run for 12 hours on three 9V batteries (the front-panel Power LED also indicates battery strength).
- Includes a carrying strap for easy portability.

MPL-1602

Rackmount 16x2 Line Mixer

The MPL-1602 provides a 16 x 2 stereo mic/line mixer in an easily installed 2RU high package. Equipped with 2-band EQ and balance and level controls on each channel, the MPL-1602 is ideal for recording, keyboard setups, broadcasting, live concert and contracting applications. It also features 2 main left/right faders, a stereo headphone jack with dedicated level control, 16 electronically-balanced 1/4˝ line inputs, 2 XLR mic inputs with trim controls, 2 pre-fade Aux sends and 2 stereo Aux returns, along with 6 channel insert and 2 main bus insert patch points. A highly practical audio tool for all kinds of professionals, the MPL-1602 is a great way to add more inputs with mixing capability to keyboard setups, small live sound systems, Techno bands and multimedia situations.

- 8 balanced stereo line input channels
- 2 XLR-balanced mic inputs with trim controls on rear panel
- Level and balance controls on each channel
- 2-band equalization on all channels
- 2 aux sends on each channel
- 2 stereo aux returns (which have independent level and balance controls allowing for use as 4 mono Aux returns)
- 2 balanced main L/R bus outs
- Front-panel stereo headphone output with dedicated level control
- 6 channel inserts and 2 main bus inserts for increased flexibility

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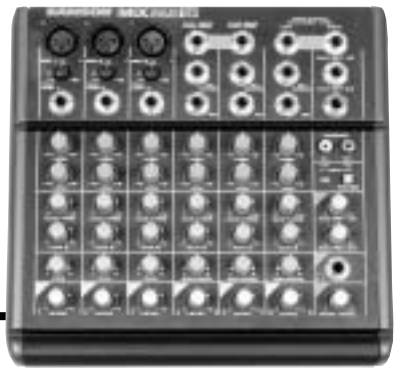
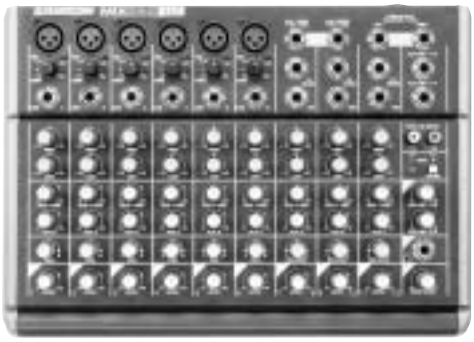
MIXPAD 9 / MIXPAD 12

9-and 12-Channel Compact Audio Mixers



Featuring a high headroom design for matchless audio performance, the deceptively sized Mixpad 9 is small but professional in every respect. Like the Mixpad 4 it offers superior low-noise and low distortion specifications along with an exceptionally low group delay over the full bandwidth for a transparent, open sound. An ideal choice for smaller live sound situations such as clubs, acoustic setups, keyboard rigs and lounge acts, project recording and more.

Remarkably compact 9- and 12-channel mixers, the Mixpad 9 and 12 offer professional audio performance and a wide range of user-intensive features for virtually any type of application. They boast superior low noise and low distortion specifications, include wide-range gain trim controls for both mic and line inputs, provide very transparent, open sound, and they have a very high slew rate—usually found on larger, more expensive mixers. They offer phantom power on all mic inputs for use with condenser mics and an in-line power supply eliminates magnetically-induced hum.



Fully professional, the Mixpad 12's complete feature set and high-quality audio performance allows it to function very effectively in the studio, sound reinforcement, field recording, audio for video and film applications and more. Almost identical except for their inputs, both feature 2-band EQ and pan/balance and level controls on each channel, a front-panel stereo headphone jack, 3 electronically balanced mic/line inputs (XLR and 1/4″ connectors), three 1/4″ stereo line inputs, a CD/tape input, 48-volt phantom power, 2 Aux sends and 2 stereo Aux returns (which have independent level controls and so can be used as 4 mono Aux returns), as well as a main level control and power and peak LEDs.

- In the MIXPAD 12, a total of twelve input channels (including three stereo channels); in the MIXPAD 9, a total of nine input channels (including three stereo channels);
- Mono channels provide electronically balanced inputs that can be used for mic or line-level input, while stereo channels are ideal for line-level sources such as outboard signal processors; CD players; tape or cassette recorders; stereo drum machines; and keyboards and MIDI tone modules.
- An electronically balanced main stereo output for connection to a power amplifier or tape recorder.
- Dedicated Tape/CD input
- Two auxiliary sends and two stereo auxiliary returns (which can be used as four monophonic returns). Aux send 1 is pre-fader (but post-equalizer), making it ideal for use as a headphone or monitor cue mix, while Aux send 2 is post-fader and post-equalizer.
- Independent 2-band EQ for each channel, with 1±5 dB of cut/boost for low (100 Hz) and high (10 kHz) frequencies.
- Constant level pan controls for placing each mono channel in the left-right stereo spectrum, as well as balance controls that let you blend the relative levels of stereo inputs.
- Mic input trims are continuously adjustable from +4 to -50 dB, making it possible to use them with a wide variety of microphones.
- Provide 48 volts of phantom power to all mic inputs
- Peak LEDs for the left and right main outputs, showing you when signal is overloading or near overloading.
- Center detents for all pan, balance, and EQ controls, making it easy to access them in low-light situations like live performance.
- Convenient front-panel Power switch, Power LED and dedicated headphone jack.

	Mixpad 4	Mixpad 9	Mixpad 12
Frequency Response	±3dB 10Hz to 50kHz	±3dB 10Hz to 50kHz	±3dB 10Hz to 50kHz
Signal To Noise Ratio	95dB	95dB	95dB
THD	< 0.01%	< 0.01%	< 0.01%
Dimensions (W x D x H)	6.4x 9 x 2.7″	9.4 x 9 x 2.3″	12.8 x 9 x 2.3″
Weight	2 lb.	5.5 lb.	6.5 lb.

SAMSON

MDR SERIES

Samson's MDR (Maximum Dynamic Range) mixer series of mixers offer transparent sound, low noise and high quality mic preamps at very affordable prices. Each mixer features low impedance circuit design, 60mm faders, combi XLR connectors, phantom power, steel construction and all but the MDR 6 have built in 24-bit DSP. Samson's exclusive Hard Disc Record Mode (MDR 6 and MDR 8 only) eliminates delay between instrument and monitor mix when recording to hard disc.

MDR 6 *6-Input Stereo Mixer*

The compact MDR 6 is a six-input stereo mic/line tabletop mixer in a portable and rugged package. Perfect for a small studio, the MDR 6 features 60mm faders, 3-band EQ, Aux send and pan or balance on each channel. There are four high quality mic preamps, an Effects return, dual 5-segment LEDs and a headphone out and mono out both with level controls. The left and right mix out also feature 60mm faders.

- 60mm faders on each channel and mains
- 3-band EQ on each channel
- Four XLR inputs with mic preamps
- Mono out with level control
- Dual 5-segment LEDs
- Aux send with effects return
- Hard Disc record mode for recording with a PC
- Headphone out with level control
- In-line power supply

MDR 8 *8-Input Stereo Mixer*

The MDR 8 is an eight input stereo mixer with 24-bit DSP featuring 60mm faders, 3-band EQ, two Aux sends and pan or balance on each channel. Included are six high quality mic preamps, two Effects returns Dual 6-segment LEDs, a headphone out and mono out both with level controls. The left and right mix out also feature 60mm faders.

Step-up features from the MDR 6—

- Six XLR inputs with mic preamps
- 2 Aux send with 2 effects return
- 24-bit DSP effects (lush Hall, Room and Vocal reverbs, Chorus and Delay)
- Dual 6-segment LEDs

MDR 10 *10-Input Stereo Mixer*

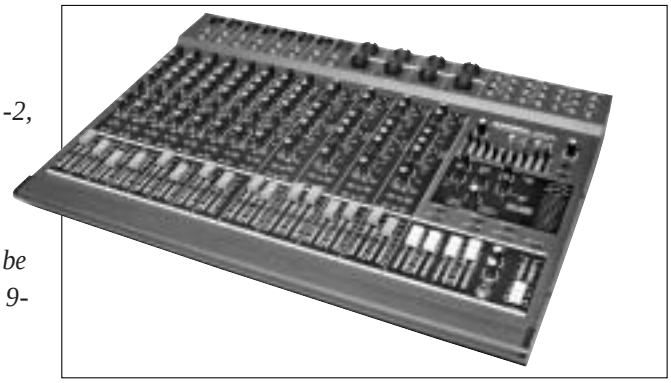
The 10-input MDR 10 packs professional features into a compact package. All 10 channels feature a high quality mic preamp, 60mm faders, a pan control and a 1/4˝ and XLR input (with mic preamp). Channels 1-6 offer a gain control, 3-band EQ with sweepable mids, a low-cut filter, two Aux sends (1 switchable pre/post, 1 to the DSP), Solo and Mute switches and smooth 60mm faders. Channels 7-10 feature 2-band EQ, 1 Aux send (switchable to Aux 1 or 2), Solo and Mute switches and 60mm faders. The outputs and bus 3 and 4 are on 60mm faders and the master section features selectable DSP, headphone out with level control, effects returns, mono out, phantom power and more.

Step-up features from the MDR 8—

- 10-channel stereo mixer with 2+2 bussing
- 10 XLR inputs with mic preamps
- 3-band EQ with sweepable mids and a low cut filter on channels 1-6
- Dual 12-segment LEDs
- I2-track in and out connectors with send to mix switch

MDR 16

16x4 Stereo Mixer



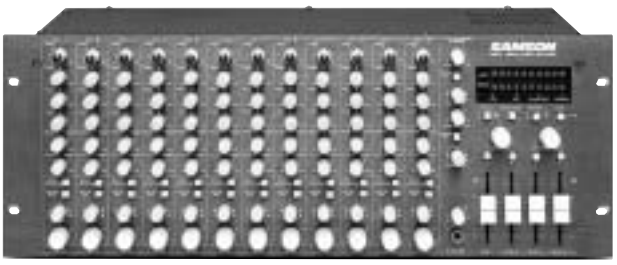
The flagship of the series, the MDR 16, is a true 4-buss mixer. All 16 channels feature a high quality mic preamp, gain control, Solo, Bus 1-2, Bus 3-4 switches and 60mm faders. Channels 1-8 offer 3-band EQ with sweepable mids, a low cut filter, two Aux sends (1 switchable pre/post, 1 to the DSP) and channel inserts. Channels 9-16 have 2-band EQ and a switchable Aux send. The four 60mm bus faders can be routed to the left/right fader. The master section features 24-bit DSP, 9-band stereo EQ, dual 12-segment metering, Solo, PFL, 2-track out, headphone and mono out with level controls and more.

Step-up features from the MDR 10—

- 16-channel stereo 4-bus mixer
- 16 XLR inputs with mic preamps
- 60mm faders on channels, busses and mains
- 3-band EQ with sweepable mids and a low cut filter on channels 1-8

MPL-1204

Rackmount 12x4 Mic/Line Mixer



"MPL" stands for "Mic/Program/Line" and the name describes the broad range of signals which can be handled by this powerful mixer. In fact, the compact design of the MPL-1204 belies its extraordinary versatility. A great all-around audio tool, the 4RU high MPL-1204 is designed for use in live and monitor mixing, keyboard setups, recording and fixed installations, conference rooms, distributed sound and as an extra mixer to expand home recording setups.
A true 4-bus, the versatile MPL-1204 features 12 input channels with mic and line trim pots, 3-band EQ, two Aux sends (1 pre, 1 post) plus solo and bussing switches. There are independent faders for each bus, and stereo Main and Control Room volume controls provide complete flexibility in monitoring and routing.

- Twelve independent channels with balanced XLR and 1/4˝mic and line inputs; also has a stereo CD/Tape input with dual RCA jacks.
- 48V phantom power for all 12 mic inputs
- Constant level Pan controls for each channel that allow you to precisely place each input signal in the left-right plane.
- Independent 3-band EQ for each channel, with 15dB of cut/boost for low (100 Hz) and high (10 kHz) frequencies, and 12 dB of cut/ boost for mid (1 kHz) frequencies.
- Channel inserts for all twelve channels, enable use of outboard signal processors such as EQs, compressors or noise gates.
- Center detents for all Pan and EQ controls make it easy to use in low-light situations.
- Flexible front panel metering system with 10-segment level meter lets you view at a glance the levels of the main Left/Right output, as well as main power, phantom power, and PFL or AFL status.
- Four bus outputs and an electronically balanced main stereo output, with dedicated front panel control. Any channel input can be routed to either of the two bus pairs (1/2 or 3/4), and any of the buses can be routed to the Left/Right mix output with the touch of a button. This flexible design allows you to easily mute selected channels and/or to create submixes within your main mix.
- Independent XLR-balanced Main stereo outputs as well as balanced stereo Control Room outputs with dedicated level control.
- Bus inserts allow external signals to be sub-mixed into any or all of the four buses and also enables linking of multiple mixers.
- Two auxiliary sends on each channel. Aux send 1 is pre-fader, Aux send 2 is post-fader.
- Two stereo auxiliary returns (route to any of the buses). with front panel level controls
- Selectable Pre Fade Listen (PFL) or After Fade Listen (AFL) soloing for each channel. Both are non-mix-destructive in that they do not affect the signal being output either by the Main, Bus, or Control Room outputs.
- Front-panel headphone output with volume control for monitoring of soloed channels.
- Mounts in any 19" rack (4RU), making it easy to integrate into existing systems.

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FOLIO NOTEPAD

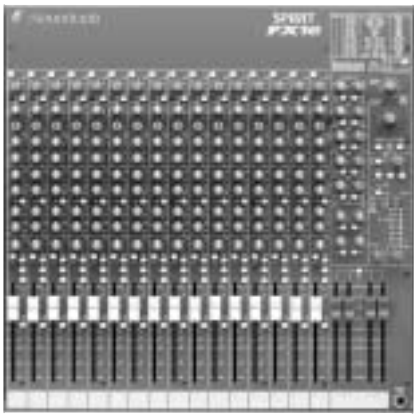
Compact Audio Mixer

The ultra-compact and very affordable Folio Notepad is packed with enough features to handle a surprisingly wide range of mixing tasks. Less than 10˝wide, it can handle up to 10-inputs, configured to give you the maximum choice of source signals, from condenser mics to turntables. Its high-quality components and advanced design ensure CD-quality sound, while its ergonomic layout makes it very easy to use. Features include separate mix and monitoring outputs, custom-built pots for precision control and carefully-tailored responses, and an external power supply that guarantees hum-free mixes with high RF rejection. Custom controls give you smooth, even response, while high quality pre-amps let you connect mic or line signals without noise or overload. Ideal for video editing, desktop recording, houses of worship, small live venues, DJ's and more.

- 10 inputs
- 4 mic inputs and 2 stereo inputs
- High quality mic preamp inputs
- 2-band EQ on all 4 mono channels
- Post-fade aux send on every input
- Stereo effect return
- Switchable +48V phantom power
- 2-track tape return
- Stereo inputs with switchable RIAA pre-amps for turntables
- Separate mix and monitor outputs
- Peak and VU metering
- Headphone output
- Custom designed rotary controls for consistent, accurate response

FOLIO FX16

16 Mic/Line Input Mixer w/Lexicon Effects



Spirit FX16 is a flexible 4-bus mixer capable of producing Digital sound quality for live and recording applications. It features a specially designed 16 program Lexicon Effects Section with dual effect capability (including Chorus and Reverb, Chorus and Delay , and Reverb and Delay) and fully editable and storable programs and parameters. The console itself includes 16 mic/line inputs and 26 inputs to mix in total (including FX returns and tape return to mix). In addition to the mix outs, two sub-buses allow groups of instruments to be sent to multitrack, to additional speakers, or sub-grouped to mix. FX16's 16 Direct Outs are individually fader pre-post switchable so they are equally useful for recording in the studio or at a gig. In keeping with the multipurpose nature of FX16, both Solo In Place and PFL solo are available, for studio monitoring and channel gain set-up applications respectively.

- All FX16's 16 mic/line inputs are equipped with UltraMic preamp which provides a full 60dB of gain range and +22dBu of headroom, meaning that the FX16 input stage can handle virtually any mic or line device.
- In the EQ section, a "truly British 3-band EQ" with swept mid benefits from custom designed pots which give greater control across carefully chosen frequencies.
- There are 4 Auxiliary Sends, including a dedicated Lexicon effects send, 1 pre-fade send and 2 pre/post-fader selectable send which are equally useful as extra foldback send in monitor-heavy live applications, or as effects sends in studio mixdown situations.
- In addition, there are four Stereo returns, a separate Mono Sum Output and 2 Subgroup Outputs.
- 100mm faders are used throughout, giving you accurate control during complex mixes, and all pots have been custom designed to give even and consistent response around their entire sweep.
- All of these features in a rugged, compact frame which can be optionally rack-mounted into a 10U space via FX16's rotating connector pod which allows leads to be connected conveniently behind the rack.

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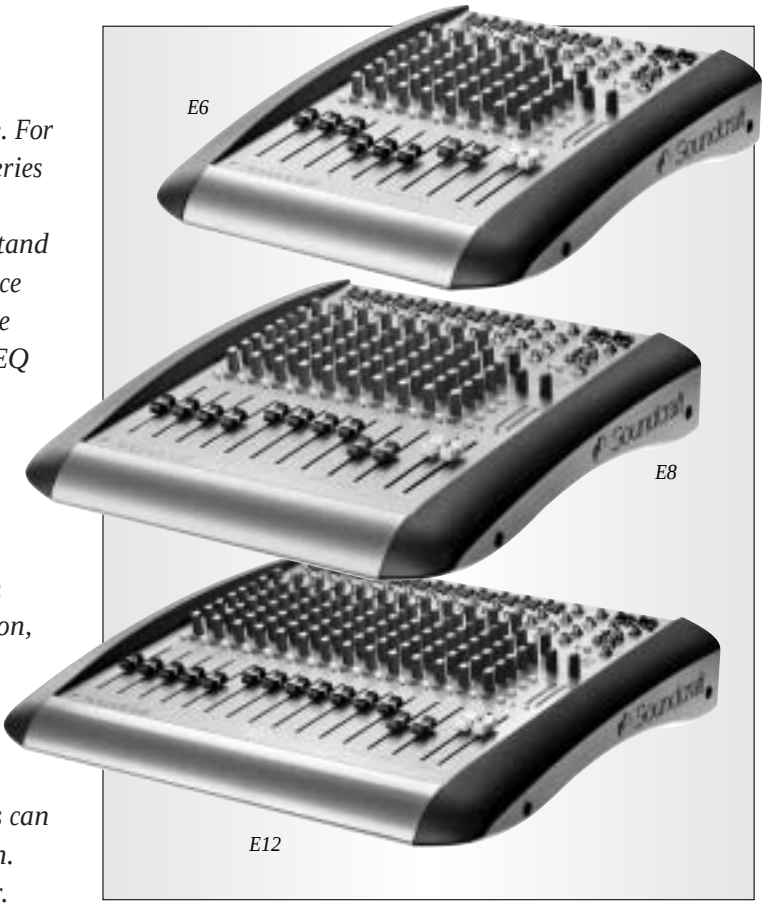
SPIRIT E SERIES

6-, 8- and 12-Input Mixers

Looking for a simple, easy to use mixer that delivers an exceptional audio performance? The Spirit E Series is here. For recording, live, install or broadcast use there's a Spirit E Series mixer for you.

The emphasis with the Spirit E Series is an easy to understand control surface uncluttered by unnecessary facilities. Surface mount technology is used throughout, using close-tolerance components for high accuracy and repeatable settings for EQ and gain controls. The mic amp features high-resolution adjustment over a wide gain range of 55dB, and provides a stunning +22dB headroom through the console. The E Series is available in 4 standard models: 6, 8 and 12 mono inputs (each model featuring two stereo inputs) and the unique ES. The ES caters for set-ups with multiple stereo sources that require simultaneous connection, such as keyboards, samplers and computers, and has 10 full-function stereo inputs as well as four mono inputs (for microphones and other mono sources). Two of these stereo inputs are equipped with RIAA - equalized inputs on RCA Phono connectors so that record turntables can be plugged straight into the mixer for DJ music production. The ES is ideal for touring keyboard players as a submixer.



FEATURES

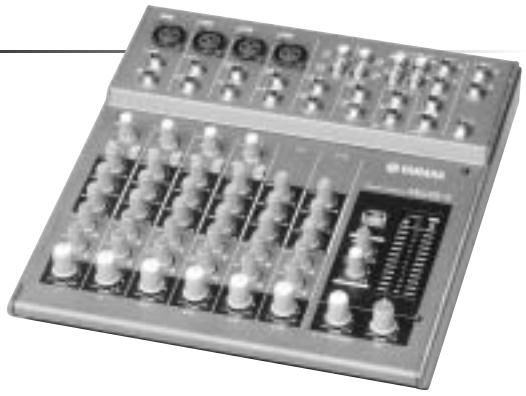
- 6, 8, and 12-mono input channel frame sizes, each with two stereo inputs
- ES Version has 10 full-function stereo inputs and 4 mono inputs
- High quality, ultra-linear mic preamps with smooth gain control
- True, professional +48V phantom power for condenser microphones
- 2 aux sends, each globally switchable pre or post fade
- All mono input channels have TRS insert sockets and inserts are also provided on the mix output.
- EQ on the standard mono input is 3-band with a swept mid, while the standard stereo input has two band EQ.
- Close-tolerance surface mount components for high-accuracy repeatable EQ
- Peak LED uses multipoint signal take-offs to watch for overload in several parts of the channel strip (post-mic pre, post-EQ, post-fade), and the LED lights more brightly as the signal approaches the peak point. .
- 100mm faders are used throughout, giving you accurate control during complex mixes
- Professional insert points for external processing
- Each mixer has 2 auxiliary buses, configurable to work the way you do, and all main connectors are professional XLR and 1/4″ metal jack sockets for reliability.
- RCA phono connectors are provided for disc and stereo playback inputs and record outputs.
- Effective grounding eliminates noise and crosstalk
- Main stereo mix has two 10-segment LED meters
- Intuitive and comprehensive solo system and a headphone output
- The monitor output and headphone output work in parallel so that performers can still listen on phones while an engineer is listening on studio monitors.
- The AC mains inlet for the integral universal switched-mode power supply is recessed underneath the console, allowing the mixers to be put against a wall with the AC cable routed under the sculptured side cheeks or mounted in a rack without the AC cable protruding on the top.
- Each mixer can be easily and quickly converted for rack mounting, simply by removing the side cheeks and adding the optional rack rails.

MG SERIES

Analog Mixers

Yamaha's analog MG mixer series cover multiple applications, including PA, project studio, classroom, house of worship, personal monitoring and sub-mixing uses. The MG-series features six models ranging in size from the small MG10/2 10-channel/2-bus unit right up to the very flexible MG32/14FX 32-channel/14-bus type with an impressive selection of built-in effects. There have been no compromises. The mixers are built for great sound, total control, and superior reliability. In fact, they undergo the same rigorous quality and reliability tests as Yamaha's world-class PM-series mixing consoles. Yet, they take full advantage of the latest Yamaha technology and manufacturing techniques to offer more value than you'll find anywhere else. In short, they offer extraordinary performance and mixing power at remarkable prices. If you need a high-performance analog mixer for music production or sound reinforcement, the Yamaha MG Series is the first — and last — place you should look.

MG10/2 Mixing Console



If you simply need to mix a few sources to stereo, but still want the finest, audio quality available, the MG10/2 is an outstanding choice. Compact and easy to use—the MG10/2 can even be mounted on a mic stand (with optional adapter) for totally flexible positioning and easy access. Ideal for demo and music production in your personal studio, for band rehearsal or small sound reinforcement applications.

10 Input Channels

- Channels 1 - 2 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 3 - 6 can be configured as two 1/4″ TRS stereo line inputs or as two mono microphone inputs.
- Channels 7 - 10 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.
- The mic preamps use high-performance head amplifiers and switchable phantom power allowing you to bring out the best of both dynamic or condenser microphones.

Aux Sends and Returns

- Two 1/4″ TRS post-fader aux sends are available for creating a monitor mix or accessing an external effects processor.
- 1/4″ TRS stereo aux returns with level control provide the means bringing an external effects unit back into the mix.

3-band Channel EQ & HPF

- Designed for smooth, "musical" response, a 3-band EQ is provided on all input channels. Mono mic input channels also feature a switchable highpass filter for cutting unwanted low-frequencies.

Mains Section

- 1/4″ TRS balanced main outputs with master level control
- 1/4″ TRS balanced control room and 1/4″ headphone outputs with level control are also provided.
- Left and right RCA record outputs as well as 2 track inputs with level control.
- 12-segment LED output meters

Optional Mic Stand Mount

- Mounts on a mic stand with the optional BMS- 10A Mic Stand Adaptor, so the mixer is always within easy reach.

MG12/4 • MG16/4 • MG16/6FX

Extensive Creative Control in the Studio or on Stage

The mid-range MG models go beyond the basics to give you extensive control for a wide range of applications - with the no-compromise Yamaha sonic quality that makes the MG mixers the finest in their class. Whether music is a hobby or profession, these mixers will deliver total satisfaction. If you don't need effects, or already have an arsenal of outboard favorites, the MG12/4 or MG16/4 may offer all the capacity and capabilities you need. But if the idea of having some of the finest effects available built right into the console appeals to you, then consider the effect enabled MG16/6FX.

Step-up Features from the MG10/2

MG12/4 – 12 Input Channels

- Channels 1 - 4 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 5 - 8 can be configured as two 1/4″ TRS stereo line inputs or as two mono microphone inputs.
- Channels 9 - 12 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

MG16/4 – 16 Input Channels

- Channels 1 - 8 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 9 - 12 can be configured as two 1/4″ TRS stereo line inputs or as two mono microphone inputs.
- Channels 13 - 16 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

Stereo Group Buss

- A stereo group bus with 1/4″ TRS outputs adds a convenient way of channel grouping – ideal for creating a discrete signal path for tracking or creating a monitor mix.

Illuminated Switches

- Each input channel features illuminated switches for ON (Stereo buss assign), PFL (Pre- Fader Listen), and phantom power providing visual confirmation of settings.
- Faders are provided for the input channels, stereo group and main stereo bus.
- Separate level controls are provided for each channel's two Aux Sends. Additionally, Aux one can be switched Pre or Post fader.
- Master Send controls for controlling output levels independently for Aux 1/2
- The main stereo outputs feature both XLR and 1/4″ TRS output connectors.
- Supplied with rackmount adapters, use them on a desktop or mounted in a rack

MG16/6FX Mixer with Effects

16 Input Channels

- Channels 1 - 8 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 9 - 12 can be configured as two 1/4″ TRS stereo line inputs or as two mono microphone inputs.
- Channels 13 - 16 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

Internal Digital Effects

- Built-in internal effects processor with 16 superb reverb and delay programs with a variable parameter control.
- An effect send on each input is provided for controlling the amount of effect for each channel.

7-band Stereo Graphic EQ

- A 7-band stereo graphic equalizer with ±12dB boost/cut per band is available for flexible overall response shaping control.

Six Buses (Stereo and Two Group Pairs)

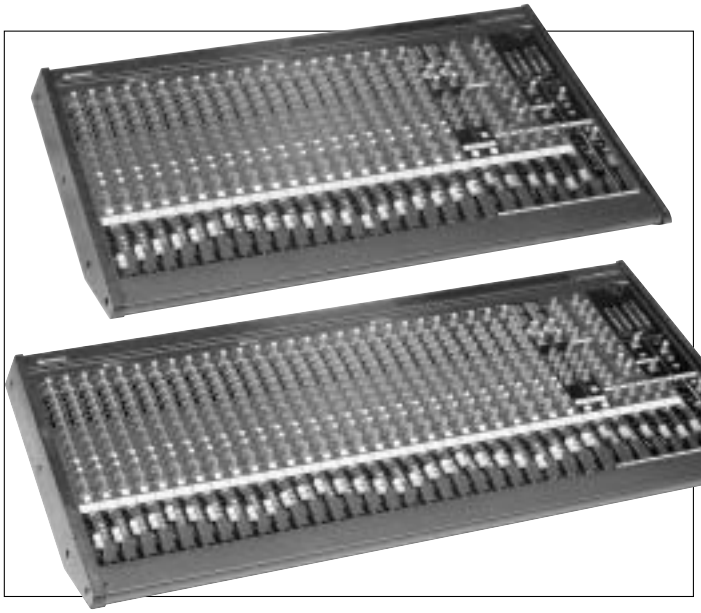
- Two pairs of stereo group buses are provided for convenient channel grouping, in addition to the main stereo bus.

Enhanced Channel EQ

- The mono input channels feature 3-band EQ with a sweepable mid band as well as a High Pass Filter for cutting out unwanted low frequencies on the mic inputs.
- Stereo input channels feature four fixed frequency bands (High, High-Mid, Low-Mid and Low).

YAMAHA

MG24/14FX • MG32/14FX



Serious Capacity for Sound Reinforcement and Installations

If your application is live sound reinforcement you'll want all the channel capacity you can get - just in case. Vocal mics, instrument mics, stereo keyboards, direct-injection feeds, drum mics, and the rest can add up very quickly. With 24 and 32 input channels, respectively, the MG24/14FX and MG32/14FX are ready to handle all but the most ambitious sound-reinforcement setups. And with dual SPX digital effect systems on-board you won't need racks of outboard gear to get the sound you need. There's also a comprehensive range of group and auxiliary busses to make even complex mixes easy.

FEATURES

MG24/14FX – 24 Input Channels

- Channels 1 - 16 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 17 - 20 are configured as two 1/4″ TRS stereo line inputs.
- Channels 21 - 24 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

MG/32/14FX – 32 Input Channels

- Channels 1 - 24 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 25 - 28 are configured as two 1/4″ TRS stereo line inputs.
- Channels 29 - 32 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

Mic Preamps

- All 16 (MG24/14FX) and 24 (MG32/14FX) mic preamps are of exemplary quality featuring low-noise, high-precision head amplifiers with phantom power switchable in 8-channel groups.

14 Buses For Flexible Signal Routing

- In addition to lots of input channels, live sound reinforcement applications usually demand a number of additional mixes - usually in the form of group sub-mixes and aux sends for external signal processing and monitor mixes.
- Four stereo group bus pairs for convenient channel grouping.
- Six auxiliary busses (four configurable for pre- or post-fader operation and two set up as effect sends).
- Two internal effect busses feed the dual high-performance built-in SPX processors.

Dual SPX Digital Effects

- Not one, but two high-performance digital signal processing stages, fed by separate effect buses, are provided using the very latest Yamaha DSP technology. Each stage provides a selection of 16 professional-quality SPX digital effects, including reverb, delay, pitch change, chorus, phasing, vocal doubling, distortion, and more.
- Parameter controls that can be adjusted to tailor the effects to your sonic requirement are also provided and Tap delay allows you to create tempo-synchronized delays.

3-band EQ with Sweepable Mid and HPF

- 3-band EQ with sweepable midband provided on all input channels provide exceptionally smooth, intuitive response, helping you create cleaner, tighter mixes.
- All mono mic input channels also feature a switchable high-pass filter that can be used to cut out unwanted low-frequency noise.

Talkback Input

- Communication capability is important for efficient setup as well as for keeping a show running smoothly. Both the MG24/14FX and MG32/14FX feature a talkback system that allows the FOH engineer to communicate with the monitor engineer, performers, or other staff to keep the team operating at optimum efficiency.

Sweepable LPF for Mono Out

- The XLR balanced mono output, aided by the built-in sweepable low-pass filter, provides an ideal way to tune and drive a subwoofer system.
- An XLR input on the top of the console is provided to accommodate a talkback mic
- Balanced XLR Stereo and Mono Outputs

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MG SERIES COMPARISON

	MG10/2	MG12/4	MG16/4	MG16/6FX	MG24/14FX	MG32/14FX
Input Channels	10	12	16	16	24	32
XLR Mic Inputs	4	6	10	10	16	16
1/4″ TRS Line Inputs	2	4	8	8	16	16
Stereo Inputs	2	2	2	2	4	4
1/4″ TRS Insert I/O	Ch 1 – 2	Ch 1 – 4	Ch 1 – 8	Ch 1 – 8	Ch 1 – 16	Ch 1 – 16
Phantom Power	+ 48 V	+ 48 V	+ 48 V	+ 48 V	+ 48 V	+ 48 V
Input Gain Control	44 dB variable	44 dB variable	44 dB variable	44 dB variable	44 dB variable	44 dB variable
High Pass Filter	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct
AUX Send/Return						
1/4″ TRS Send	2 (1/Pre, 2/Post)	2 (1/Post-Pre, 2/Post)	2 (1/Post-Pre, 2/Post)	2 (1/Pre, 2/Post-Pre)	6 (1-4 Post-Pre, 5-6 Post)	6 (1-4 Post-Pre, 5-6 Post)
1/4″ TRS Stereo Aux Return	1	1	1	1	2	2
Effect Send	—	—	—	1	2	2
2 Track In	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA
Record Out	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA
Main Stereo Out	1/4″ TRS	1/4″ TRS & XLR	1/4″ TRS & XLR	1/4″ TRS & XLR	1/4″ TRS & XLR	1/4″ TRS & XLR
Main Stereo Inserts	—	—	—	—	1/4″ TRS	1/4″ TRS
Stereo C/R Out	1/4″ TRS	1/4″ TRS	1/4″ TRS	1/4″ TRS	—	—
Stereo Sub Out	—	—	—	—	1/4″ TRS	1/4″ TRS
Mono (Sub) Out	—	—	—	—	XLR	XLR
Mono Out Low Pass Filter	—	—	—	—	80-120 Hz 12dB/Oct	80-120 Hz 12dB/Oct
1/4″ TRS Group Out	—	2	2	4	4	4
1/4″ TRS Group Inserts	—	—	—	—	4	4
Stereo Headphone Output	1/4″ TRS	1/4″ TRS	1/4″ TRS	1/4″ TRS	1/4″ TRS	1/4″ TRS
CH EQ (Mono Channels)						
Boost/Cut per band	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)
High (Shelving)	10 kHz	10 kHz	10 kHz	10 kHz	10 kHz	10 kHz
Mid (Peaking)	2.5 kHz	2.5 kHz	2.5 kHz	0.25 – 5 kHz	0.25 – 5 kHz	0.25 – 5 kHz
Low (Shelving)	100 Hz	100 Hz	100 Hz	100 Hz	100 Hz	100 Hz
CH EQ (Stereo Channels)						
Boost/Cut per band	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)
High (Shelving)	10 kHz	10 kHz	10 kHz	10 kHz	10 kHz	10 kHz
Hi-Mid	—	—	—	3 kHz (Peaking)	3 kHz (Peaking)	3 kHz (Peaking)
Mid (Peaking)	2.5 kHz	2.5 kHz	2.5 kHz	—		
Low-Mid	—	—	—	800 Hz (Peaking)	800 Hz (Peaking)	800 Hz (Peaking)
Low (Shelving)	100 Hz	100 Hz	100 Hz	100 Hz	100 Hz	100 Hz
7-band Graphic Equalizer	—	—	—	(125, 250, 500, 1 k, 2 k, 4 k, 8 kHz)	—	—
Boost/Cut per Band	—	—	—	±12 dB (Max.)		
Internal Digital Effect	—	—	—	1 (16 Programs)	2 (16 Prgs each)	2 (16 Prgs each)
Power Supply	External	External	External	External	Internal	Internal
Dimensions (WDH)	10.2 x 11.3 x 2.5″	12.5 x 16.4 x 4.1″	16.7 x 16.4 x 4.1″	16.7 x 16.4 x 4.1″	33.5 x 21.3 x 5.9″	41.7 x 21.3 x 5.9″
Mounting Capabilities	Mic Stand Mountable	Rack Mountable	Rack Mountable	Rack Mountable	—	—
Option Mic Stand Adapter	BMS-10A	—	—	—	—	—

SONIC SPECIFICATIONS	
Total Harmonic Distortion	Less than 0.1 % (THD+N) 20 Hz – 20 kHz @ +14 dB 600 Ω (ST OUT)
Frequency Response	0 +1, -3 dB 20 Hz – 20 kHz @ +4 dB 600 Ω (ST OUT)
Input Hum & Noise	128 dB 20 Hz – 20 kHz, Rs=150 Ω, Input Gain=Max, Input sensitivity=-60 dB
Crosstalk	-70 dB @ 1 kHz

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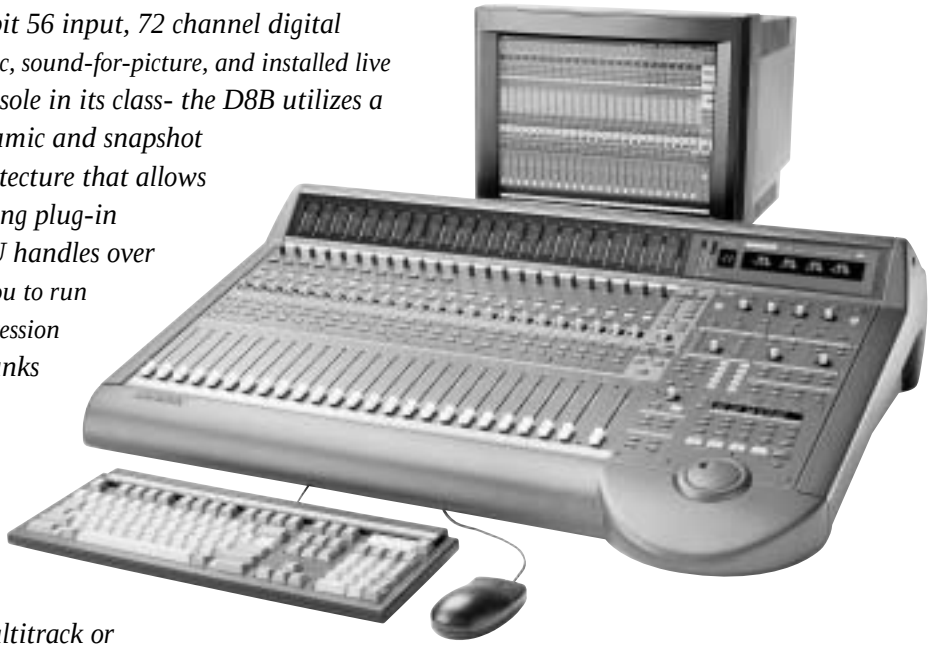
D8B (DIGITAL 8•BUS)



The Mackie Digital 8 Bus or D8B is a 24-bit 56 input, 72 channel digital mixing console designed for professional music, sound-for-picture, and installed live applications. The most advanced digital console in its class- the D8B utilizes a separate CPU for controlling limitless dynamic and snapshot automation capabilities and an open architecture that allows software upgrades as well as signal processing plug-in expansion. The pentium-based 32-bit CPU handles over 3 billion instructions per second enabling you to run digital functions such as EQ, gating and compression on forty-eight simultaneous channels. And thanks to the Mackie Real Time OS the system doesn't get bogged down by another operating system such as the Mac OS or Windows. The addition of an SVGA monitor, keyboard and mouse make working the D8B a dream. Whether you are working with analog tape, a digital multitrack or DAW, optional multi channel I/O cards are available to ensure that the D8B will integrate seamlessly into your system. A dedicated DSP processor provides two stereo effects and up to three additional DSP cards can be installed for even greater signal processing power. The D8B's smart ergonomic design, digital patchbay and surround sound mixing capabilities as well as the studio grade 24-bit converters and Apogee UV22 16-bit CD encoding system far exceed the flexibility and sonic integrity of anything else on the market.

FEATURES

- 56 inputs, 72 channels
- Full big-console automation of all parameters including digital effects—recall and edit any mix anytime
- 48 channels of audio with gating, compression and parametric EQ available on all channels simultaneously. 48-channel overview screen
- 12 balanced XLR-mic inputs with switchable phantom power, 1/4″ TRS line inputs with high quality mic preamps and 1/4″ TRS inserts. There is an additional twelve 1/4″ TRS balanced/unbalanced line inputs
- Channels 25 to 48 are accessed from three card slots that accept any combination of optional 8 channel I/O cards
- 12 Aux sends, AES/EBU and S/PDIF digital I/O, 8-channel sub/ surround outputs (25-pin D-sub connector), 2 stereo monitor outputs, balanced XLR and 1/4″ master outs and 3x 1/4″ balanced 2-track returns.
- Built-in meter bridge and Talkback mic
- Speaker select and level controls let you choose between two sets of monitors (eg. near and far-field) as well as two independent, automatable Phones/Cue Mix sections.
- Insert and patch point feature on channels - source either plug-ins, 8 aux sends, 8 buses or 72 channels (pre-DSP only)
- Multifunction display provides tape position in SMPTE /MIDI and lets you mark multiple in and out points.
- Dynamic and snapshot automation
- Event automation track, graphic time line and Event list style automation editing
- MIDI Machine Control
- Assignable MIDI commands on transport and on the D8B master control section
- 999 levels of undo
- Up to 999 samples of time offset (delay) for each channel pre-DSP (dynamics/EQ)
- Inserts can be run across Mains and Busses
- Open architecture compatible with 3rd party plug-ins. Up to 16 simultaneous plug-ins on the first 48 channels, pre or post DSP, pre-fader via up to 4 UFX cards
- Cross patching allows channels from various banks to be substituted with those from other banks
- Enhanced dynamics with input keying and EQ filter, soft-knee compression toggle, linear and exponential option
- Advanced Mix Editor with view sizing arrows, auto-punch, auto-loop SMPTE time code boxes, Loop-in/out, Punch-in/out and Locate markers in the time bars
- Drag and drop file management between File windows and desktop
- Auto Punch can be toggled from the console via MMC.
- 5.1, 6.1 and 7.1 surround-sound capabilities in an adjustable 3-D floating window.

D8B (DIGITAL 8•BUS)

Channel STRIP

- Built-in meter bridge displays the selected channel's input level, including level display for tracking, mixing, effect returns, and sub bussing.
- Trim Level (channels 1-24 only) with quick and easy level setting
- Mic/Line Switch (channels 1-12).
- Record Ready button on each channel strip lets you remotely arm up to 24 tracks of your digital multi-track or hard disk recorder (on recorders that support MMC) without turning away from the console.
- Assign button lets you quickly route channels to buses or any of the 24 tape outputs with the punch of a button or a click of the mouse. It's just like a digital patch bay!
- Write button displays the automation state on the channel strip. For those who want hands-on ability, the D8B's automation can also be run in Auto-Touch mode. As soon as any recordable control is touched and adjusted, the corresponding channel automatically goes into record mode.
- V-Pot gives you hands-on control over all channels' panning as well as all auxiliary send, tape output, and digital trim levels. V-Pots have visual feedback, instant recall and full automation capability.
- Bank LED indicator tells you whether the channels are currently Mic/Line, Tape In, or Effect Return channels. The bank LEDs switch along with the selected fader bank.
- Select function for channel editing, grouping, sending to tape outputs, linking, group or bus assignment, or clipboard functions.
- Direct access Solo and Mute buttons—no scrolling through menus
- 100mm motorized channel faders are silky-smooth when you move them; jitter-free in motorized mode. The motorized action can be globally defeated.

Apogee UV22 Super CD Encoding

Built into the D8B is Apogee's UV22 outboard processor which encodes more bits of detail onto CDs. Audible benefits include more detailed reverb, better tonal balance, wider stereo spread, more precise placement of instruments in the sound stage, and lower noise. Produce fantastic-sounding 2-track masters that will continue to sound fantastic no matter who masters and duplicates them.

Fat Channel – For Hands-on Processing

Perhaps one of the favorite features of the D8B is the legendary Fat Channel. And now it has a new look and many new features that will greatly augment your audio arsenal. Larger views, adjustable dynamics EQ key types, Gate expansion, and external MIDI gate trigger are just a few of the new Fat Channel components that will make your mixing tasks easier and more effective.

- Provides four V-Pots and Previous/Next scroll buttons to quickly find and adjust effects parameters of all twelve auxiliary sends and 4-band EQ, compressor with limiting, and gate with expansion mode parameters for 48 channels
- The Compressor and Gate parameters can be viewed graphically in the expanded dynamics view windows.
- The Dynamic Key provides a selection of four EQ types (Parametric, Shelf, High Pass or Low Pass) that can be applied to the Key input.
- A bright vacuum fluorescent display above the FAT Channel gives you visible read-outs of changing DSP values. (In addition to the optional SVGA monitor.) Although you can tweak the D8B's settings with its analog-style knobs, you may prefer the on-screen control panels for intricate adjustments. To quickly check channels for phase inversion, you can flip the phase of individual channels from the VFD.

Master Section

- Lurking behind the Digital 8•Bus' 100mm motorized faders are four banks of channels, each available at the touch of a dedicated button.
  - —Channels 1-24 are for tracking
  - —Channels 25 to 48 are for tape returns
  - —A third bank for Internal Stereo Effects Returns and four External Stereo Effects Returns
  - —A fourth bank - 8 virtual group master faders, 8 dedicated MIDI controller faders, and 8 bus master faders. When you switch banks, all faders will "snap" to their relative position in the selected fader bank. The SVGA display changes, too, reflecting the selected bank and all controller positions.
- Choose between near-field or main speaker outputs to monitor your mixes. This section can receive source signal from your stereo mastering returns via three sets of analog two-track inputs and/or the stereo AES/EBU or S/PDIF input. Plus, a built-in 'Talkback' mic with 'Dim' button allow communication with studio talent through the Phones/Cue mixes or Studio outputs.
- Each provides an amplified headphone output of either the Control Room mix or either of the 'Cue Mixes' (Aux 9/10, 11/12)
- A unique 'Copy Mix to Cue' function allows you to set up your headphone mixes using the D8B faders, and then copying their relative positions to the Cue Mixes.

External CPU

The D8B includes a separate, external CPU with Pentium processor, complete with 32 MB of RAM, large capacity hard disk, 3.5″ floppy drive (use to load additional software or save sessions), and SVGA video card, keyboard and mouse connections for connecting an optional SVGA monitor (for display of many console functions including fader banks, DSP and effects, surround sound, software library, and file management), keyboard and mouse. On the back is an Ethernet connector. Use it to cascade two D8B's solo control logic, mutes, transports, etc… or connect one or more Digital 8•Bus's to Mac's or PC's via a LAN for transfer of session files and EQ/Effects libraries.

D8B (DIGITAL 8•BUS)

Locator/Transport/Jog

- Once you start using it, you'll discover that you don't have to take your eyes off of the console nearly as much. You can run the transport of any tape or hard-disk based recorder that can slave to MIDI Machine Controlled. You can create song start and stop or loop continuously between two points. And you can dial in any location point in a session right from the console.
- You can also arm up to 24 tape machine tracks directly from the Record Ready button on each D8B channel strip. The time code counter window and location point reader above the transport section reflect the tape position in SMPTE (hrs:mins:secs:frms) or MIDI (bars:beats:ticks), and let you mark multiple location points.

Easy Surround Sound

The future of music is multichannel and the D8B makes it easy. To move a channel from front to back or side to side, just pull up the Surround window and use the mouse to "grab" the sound source and move it around in the Pan window. You can select two different positions for a sound and then "morph" it from one spot to another over a specified length of time. There is also a mind-boggling Surround Screen that shows the front-back/left-right position of up to 72 different channels—all in one view.

- Surround Sound mixing environment with depth of center control, surround LFE gain control for each channel, sound-corrected buss and track assignment.
- Surround front-to-rear pan control via control surfaces or MIDI
- Pan Depth of Center for keeping sound effects and/or music out of the center speaker during LCRS, 5.1/6.1/7.1 sessions.
- The Surround Sound window includes nine faders (eight surround and a master) for adjusting the analog output levels at the Bus 1-8 outputs. This doesn't affect the digital surround output signals at the Alt 1-8 outputs. This allows you to make adjustments in your monitoring levels without affecting the signal being recorded. Once the levels are set, you can lock the monitor level settings by clicking the Menu button and choosing Lock Monitor Levels.

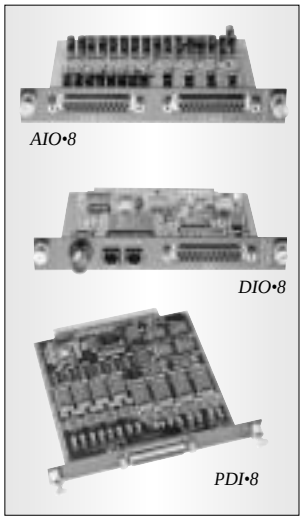
HUI Mode

The D8B software includes a HUI mode, which allows the D8B to operate as a HUI (Human User Interface) control surface with DAW software applications that support the Mackie HUI. Turn on HUI mode by clicking Options in the top menu bar and selecting HUI Mode. This creates a fifth fader bank in addition to MIC/LINE, TAPE IN, EFFECTS, and MASTERS, which can be accessed by pressing SHIFT+MASTERS on the control surface or clicking the HUI button above the Master fader on-screen. HUI mode uses the MIDI IN/OUT ports connected to a MIDI interface to control specific functions in the DAW, including fader and transport control.

Open Card Cage Architecture

The D8B can be configured for almost any recording application. It offers analog I/O, digital I/O, word clock I/O and effect processing cards as add-on options. And it doesn't take a degree in electrical engineering to add a card. In fact, you don't even need a screwdriver to install a card.

OPTIONAL 8-CHANNEL I/O CARDS



AIO•8 Analog I/O Card:
8 balanced line level (+4dBu) outputs and returns for connecting the D8B to analog equipment.

DIO•8 Digital I/O Card:
Made by Apogee, this card gives you 8 channels of 24-bit digital tape outputs and digital tape returns in both ADAT lightpipe and Tascam T/DIF formats. The DIO•8's provide direct dubbing capabilities from ADAT optical to T/DIF format, and vice versa.

OPT•8 Low-Cost Digital I/O Card:
The low-cost OPT•8 I/O card provides 8 channels of ADAT lightpipe for under $100!

PDI•8: 24-bit AES/EBU Digital I/O Card;
Ideal for digital audio workstations and supports real time sample rate conversion on input as well as Apogee's UV22 16-bit CD encoding on output.

Apogee Word Clock I/O
If you need to sync your D8B to an external clock reference, the optional Apogee Word Clock I/O card provides a BNC clock input for slaving the D8B. Extremely low-jitter, cost-effective clocking card with vari-speed and pull-up/down capabilities.

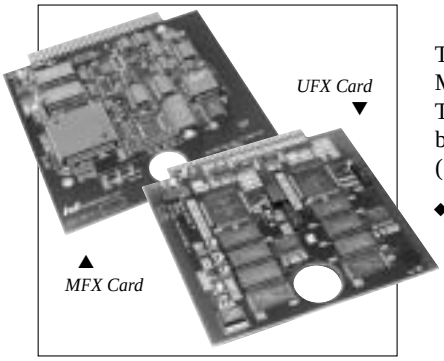
D8B (DIGITAL 8•BUS)

Mackie MFX and UFX Cards and Digital 8•Bus Plug-ins

The D8B's open architecture lets you take advantage of some of the world's most creative software companies. These plug-ins are made possible by the D8B's advanced operating system and optional MFX and UFX cards. The D8B has four card slots in the rear labeled "Digital Effects Cards". The D8B comes with one MFX card comes installed. That leaves three slots to fill up. You can mix and match MFX and UFX cards depending on the plug-ins you want to use. The D8B is the only digital consoles that supports such a wide variety of plug-ins from the world's best processor and effects companies and there's more to come.

MFX Card

The MFX card is the host for Mackie Effects reverbs, choruses, echos etc... The IVL Vocal Studio Demo is included. Adding a second MFX card unlocks the official Vocal Studio plug-in. (One MFX card comes with your D8B.) MFX cards can be mixed with UFX cards, up to a total of four. Removing the MFX card will disable the Mackie Effects, and Vocal Studio.



UFX Card

The UFX Card is the host for Acuma, Antares, Drawmer, and Massenburg plug-ins. Mackie Mono Delay and T.C Electronic's TC FX II (provides reverbs from the renowned M2000) are bundled free with a UFX card. UFX and MFX cards can be mixed (up to a total of four cards).

- UFX cards are 4-in/4-out, which means each card can simultaneously support either: Four mono effects, or two mono and one stereo effect, or two stereo effects
- Up to four UFX cards can be installed in the D8B. This allows you to run up to 16 channels of mono or 8 channels of stereo plug-in effects simultaneously.

Antares
AUTO-TUNE

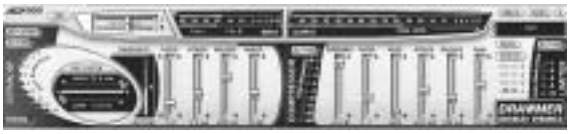
Pitch Correction for Voice or Solo Instrument

The worldwide standard in professional pitch correction, Auto-Tune corrects intonation problems in vocals or solo instruments, in real time, without distortion or artifacts, while preserving all of the expressive nuance of the original performance. In fact, Auto-Tune offers audio quality so pristine that the only difference between what goes in and what comes out is the intonation.

The precision by which Auto-Tune detects pitch is extraordinary. At a frequency of 400 Hz and a sample rate of 44100, the Auto-Tune DSP algorithm computes the pitch to an accuracy of .0001 samples per cycle, or .0004 Hz. At this resolution, the very question "What is pitch?" becomes relevant. That is, as the pitch of typical performances continuously change, the amount of variation in pitch, even over the time of a few cycles, changes greatly in comparison to the accuracy by which Auto-Tune computes pitch.

Drawmer
ADX100

Frequency-Conscious Gating, Compression, Limiting and Expansion



Acknowledged as master of the analog dynamics processor, Drawmer now offers its expertise to the Mackie Digital 8•Bus.

- Left and Right input levels. -15.75dB to +16dB in 0.25dB increments.
- Input VU meters. -42dB to 0dB.
- Faders for quick parameter control. All thresholds and levels are in linear dBs, but other controls are scaled to a user friendly law which minimizes 'fine tuning' and allows a wide range of operation.
- You can instantly recall two preferred settings.
- Noise Gate is a switch designed for percussive signals, such as drums where it can shape the envelope to create a crisp, well defined signal.
- Gate/Duck switch. When Duck is selected, the operation is reversed, so that when the signal is above threshold, the output is attenuated. Below Threshold, the signal passes unattenuated.

ACUMA LABS PLUG-INS for the D8B

A division of Mackie, Acuma Labs Ltd. develops real-time embedded systems for professional audio applications, specializing in digital signal processors, micro processors, digital audio effects, analog and digital software, real-time operating systems, interfaces, and hardware design. Seven plug-ins altogether, all of their parameters are automated through the D8B's automation engine and can be stored or recalled as a preset.

DSR-1 *Three-band Frequency-Controlled "DeEsser" Dynamic Processor*

The DSR-1 is a highly accurate, frequency controlled, three-band dynamic processor that enables you to quickly isolate and correct unwanted sibilance found in vocal recordings of singers and speakers. The DSR-1 reduces annoying sibilance and popping sounds that are often found in recordings without loosing the crisp top-end clarity. Don't let the simple chicken head controls and the funky retro-look fool you; the DSR-1 is a highly professional tool that is ideal for editing vocals, instruments, and other sources. The DSR-1 is an invaluable D8B plug-in that can save otherwise great recordings by eliminating troublesome "Ess" frequencies.

The DSR-1 employs frequency controlled compression to reduce problematic "Ess" that often occurs in recordings of singers and speakers. Using the Frequency and Width controls to zero in on specific problem areas, an internal keying device known as Listen allows you to solo or (Listen) to the isolated sibilant frequency that you want to compress. The DSR-1 can also help to reduce shrill, high-pitched sounds that may accompany bass guitars, wind instruments, or other acoustic instruments. With a series of useful presets designed for male and female vocals, spoken word and other instruments, the DSR-1 promises to make even the toughest job easy to correct.

FINAL MIX

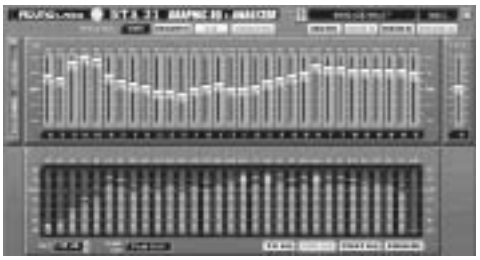
Real-time Stereo Mastering Processor

Now master your sessions within the D8B, printing directly to hard disk without having to rely on expensive mastering houses or outboard gear. Final Mix will dramatically elevate the quality of your mixes and help you create your own professional masters.

- 6-band pre-dynamics parametric EQ, 3-band dynamics processor, 6-band post-dynamics parametric EQ
- Graphical (user definable) dynamics band contours
- Adjustable crossover points and slopes for multi-band dynamics
- Node based adjustment of EQ bands and dynamics bands
- Soft Clip feature provides peak overload protection
- Noise gate with threshold adjustment
- DC removal filter automatically removes DC offset noise
- Dynamics bands linking – simplifies use in stereo configurations
- Fully automated; Global enable button
- On/Off for the dynamics section and each EQ
- Separate dynamics and EQ reset buttons
- Plug-in patch load and save
- Memories A and B – compare settings quickly and easily
- Input, output and gain reduction metering

RTA-31

Real Time 31-Band EQ and Spectral Analysis



RTA-31 is both a Graphic EQ and a Spectrum Analyzer that enables you to easily modify the frequency response of any given signal within the D8B or from external live and studio acoustic environments. Quickly view your sound using the on-board spectrum analyzer to zero in on the troublesome highs and lows of even the most difficult rooms. The amazing 'ToEQ' feature lets you immediately identify troublesome frequencies using the Frequency Analyzer while the 31-Band Graph automatically compensates for the highs and lows of your signal.

- Use your mouse as an on-screen paint brush to quickly perform various EQ editing functions.
- Use the 1/3 Octave Spectrum Analyzer (20-20K) in live and studio applications to identify the nature of an acoustic environment.
- Selectable dB reference level (0dB to -60dB) allows you to focus in on the most important range of your signal.
- Factory presets feature useful EQ and Analyzer setups
- Selectable graph types, choose from Bar, Peak Hold, Point, Peak Difference, and Peak Bar for viewing preference.
- Control graph response time to identify quick transients.
- Immediately freeze the analyzer screen to view detailed information.

ACUMA LABS PLUG-INS for the D8B

TIME PAK

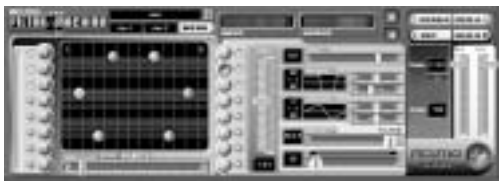
Easy-to-Use Time Modulation

An easy-to-use time modulation offering all of the classic time effects including Chorus, Flange, Auto Pan, Tremolo, combined with radically new and Xtreme settings.

- Use graphic editing sliders to adjust Level, Regen, Pan and Time Modulation, and 4th order High and Low pass filtering.

FILTER MACHINE

Intuitive, Powerful, Sweeping Filter Effects



Acuma's classic, analog styled 4-pole filtering plug-in is an easy and intuitive way to achieve powerful, sweeping filter effects.

- Select from the graphic editing sliders to adjust Pan Modulation, Frequency Modulation, and 4th order High and Low pass filtering.

- Utilizing stereo input and output paths, Time Pak's and Filter Machine's large intuitive display field lets you easily drag and drop any of the 16 nodes (Time Pack) or 16 filters (Filter Machine) to simultaneously adjust Time (Time Pack) or Frequency (Filter Machine) and Output Pan.
- Link left and right nodes (Time Pack) or left and right filters (Filter Machine) together into stereo pairs or fine tune the parameters further using 2 separate LFO blocks that incorporate preset waveforms and envelope followers with adjustable rate, depth, and phase (and volume on the Filter Machine).
- Dial in the beat of the music by selecting the BPM, or tapping the tempo in real time, and quantize the LFO rates to the nearest interval selecting from note, whole note, triplet, etc. All of the filter rates snap to the correct interval with just a click of your mouse.

MASSENBURG DESIGNWORKS

MDW2x2

2x2 High-Resolution Parametric EQ

Headed by George Massenburg, the preeminent designer, MDW (Massenburg DesignWorks) brings you the 2x2 High-Resolution Equalizer which features oversampling and double precision processing for unprecedented clarity and accuracy. Configure the MDW2x2 as two mono 5-band or one stereo 5-band parametric EQ.

DELAY FACTOR

16 Tap Stereo Delay with Modulation

Setting up a stereo, ping-pong, or multi-tap delay has never been easier! Delay Factor has a large, intuitive display field which allows you to drag and drop any of the 16 delay taps, simultaneously editing both the delay time and the output panning.

Fine tuning the delay to match the beat of the song is even easier; just dial in the BPM or tap in the tempo, and select the interval you want to quantize to - note, whole note, triplet, etc. All taps will simultaneously 'snap' to the correct interval.

Adjusting all the other parameters is easy too. All values for a single tap are automatically displayed on easy to adjust sliders when you click on the tap in the display field. Set the level, regeneration, regeneration panning, or the tap's filter. Each tap has a 4th order high and low pass filter.

SATURATED FAT

Mono Distortion & Cabinet Modeling with 7-Band EQ

As its' name implies, Saturated Fat is a mono plug-in that employs a proprietary technology to achieve warm, natural-sounding tube distortion and cabinet modeling. Choose from over 40 preset distortion types to create your "perfect sound" or select from the array of factory presets to find a setting that suits any style of music. And don't stop with guitars — Saturated Fat is great on drum tracks too! Combine the Cabinet Modeler, selecting from a list of small, medium, large, acoustic and bass amps, with the Drive Level and the on-board 7-Band EQ to build custom distortions that can be saved and recalled as User Presets. No more (loud) amplifier setups to contend with, just plug any instrument into the D8B and start playing! Saturated Fat's distinctive distortion types and amplifier models will give you killer sounds instantly.

DMX-R100

Digital Audio Mixer

A professional mixer inheriting the control philosophy of the world-famous Sony OXF-R3 Console, the DMX-R100 is a compact, 48-channel mixer with a comprehensive feature set that includes 25 motorized faders, a sophisticated control panel with touch-screen control, a fully integrated package of automation, a digital routing matrix and machine control. The DMX-R100 delivers excellent sonic performance via its state-of-the-art processing technology, offering 24-bit quality and the ability to operate at both standard and double sample rates (44.1, 48, 88.2 and 96 kHz). With its stunning sound performance, operability and flexibility, the DMX-R100 is ideal for producers, artists and engineers in applications ranging from music studios to post production and audio pre-mastering.

FEATURES

Control Panel Ergonomics

- Although the DMX-R100 is a highly cost effective mixer, it has a fully professional control surface with dedicated control knobs and switches for each individual parameter – emulating the best of traditional console ergonomics. A considerable amount of space is allocated to individual controls for fast, accurate, adjustment and they are laid out in a logical manner that reflects the way that they are used.

High Quality Sound Processing

- The DMX-R100 is designed to deliver the benefits of higher resolution audio signals – greater dynamic range and higher bandwidth. All appropriate inputs and outputs are 24-bit and both standard and double sampling rates are supported.
- The DMX-R100 also processes the full 24 bits of its digital AES/EBU I/Os without any truncation. Internal processing uses precise floating-point calculations to maintain the console's excellent sound quality.

Channels, Returns and Buses

- The DMX-R100 provides 48 input channels and 8 Aux Returns, making a total of 56 channels available for stereo or surround sound mixdown. These channels can be routed to the 8 MTR buses, 8 Aux Send buses, Master L/R Buses or Solo/PFL Buses. As well as EQ and dynamic processing for all 48 input channels, the PGM, Aux Send and MTR outputs also have EQ and dynamics.

Sophisticated Channel Faders

- 25 touch-sensitive motorized faders (24 channel and one PGM fader). The 24 channel faders can be switched in three layers; two layers for the 48 input channels and one layer for master control, including MTR masters, Aux Send masters and Return inputs. The faders are designed to give the operator a very professional 'feel'. Their 10-bit resolution provides precise level adjustment, as well as smooth and accurate replay of automation moves. he The DMX-R100 uses touch-sensitive faders because they allow for excellent operator control of automation and level.

Color SVGA Touchscreen

- The 800 x 600 color SVGA LCD touch screen provides high-quality graphics pages accessed via an intuitive menu structure. These graphics pages include channel processing, input/output routing, automation and mixer setup, and others. For example, the Channel pages gives simultaneous view and control of any one of the 48 channels. Another page, 'AUDIO OVERVIEW' gives a clear view of all 48 channels on two pages so that the operator knows instantly how the mixer is setup. The knobs, buttons, switches and LEDs are displayed on the touch screen with their size and position corresponding to the size and positions of the real controls on the control panel section. This links the visual information provided on the touch screen with the physical controls.
- In addition to numeric indications, large and clear graphic representations of the EQ and Dynamics curves are displayed on the touch screen. Additionally, the graphic/touch screen combination lets you 'zoom in' on specific control panel areas.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

DMX-R100

Channel Strip

The 24 fader strips of the DMX-R100 combine the familiarity of traditional console design with additional features derived from the OXF-R3 console. By default, each fader controls the channel gain, but Select to Fader buttons allow them to control 10 additional level adjustments in the channel path, including the eight Aux Sends, I/P Trim and MTR Sends. Similarly, the Pan rotary control defaults to stereo mix pan but can be switched to provide the same 10 level control functions as Select to Fader. As different signal paths are selected, the faders automatically move to the correct position and a ring of LEDs around each Pan control indicates its current value. This arrangement of two level controls per fader strip provides a simple method of offsetting various channel levels – one of the most common adjustments made during any audio production.

Channel Link Control

Link control of parameters such as Trim, Delay, EQ/Filter, Dynamics, Channel Cut, Channel Fader, MTR Fader, and solo mode. In normal mixing, adjacent odd and even channels can be linked in stereo pairs, and in the case of surround mixing, operators can choose to link groups from the following combinations: 1-6 , 7-12, 13-18, 19-24 channels, L/C/R, LS/RS and so on. In addition, a 'mask' function permits selective exclusion of parameters from the link operation available separately for stereo and surround channels. Surround link groups such as L/C/R, LS/RS and Sub can be selectively linked for surround mixing.

Copy and Zero Reset Function

Copy function allows channel settings of a source channel to be copied to any number of destination channels. The 'mask' function permits selective exclusion of parameters from the copy operation, and moreover, the DMX-R100 supports copying of channel fader mixes to MTR fader/Aux Send faders as well as MTR fader mixes to Aux send faders. This is extremely helpful when creating cue/foldback mixes. Additionally, a zero reset function selectively resets all level controls, faders, and knobs to their default values. EQ curves can also be set flat, and dynamics setting are set to their default values.

Flexible Internal Routing Matrix

The DMX-R100's is its internal audio routing matrix . This provides comprehensive crosspoint switching for virtually every input and output, and avoids the need for a costly external patch bay. The input section of the matrix allows any input signal to be routed to any channel. The same input signal can also be routed to multiple channels. Similarly, the output section allows bus signals to be assigned to any output including those on the four I/O slots and also allows the same signal to be assigned to multiple outputs.

The routing matrix is controlled by two touch-screen pages, one for input signal assignment and one for output signal assignment. Both pages have two levels of access. The first level provides free assignment of inputs and outputs on an individual basis. The second level supports logical groupings of inputs and outputs. For example, the block of signals from one of the optional input boards can be assigned to a group of channels. Similarly, logical groups of mixer buses, Aux Sends for example, can be assigned to a range of mixer outputs. Using this second level to work with these logical groups enables the matrix to be quickly set up. Input and output matrix crosspoint assignments are stored in snapshots. This means that a DMX-R100 can very quickly be reset to different projects by recalling snapshots that include these settings.

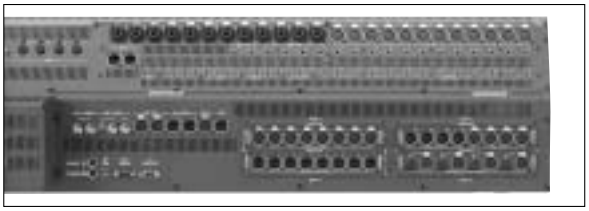
Advanced Snapshot and Dynamic Automation

The DMX-R100 includes 99 snapshots per title, making it possible to memorize and recall the state and values of virtually all mixer functions, including input matrix routing, Delay, Phase, Trim, Input Mode, Filter, EQ, Dynamics, Pan Assign, Cut, Fader and Aux.

The DMX-R100 also offers a ' library' , which is intended for storing repeatedly used EQ and Dynamics settings. Up to 99 can be stored in the library per title for later recall and assignment to any individual channel. In addition, there is comprehensive dynamic automation of Faders, Cuts, Pans, EQ and Filters, Dynamics, and Aux sends. Dynamic automation can be synchronized to both SMPTE and MTC (MIDI time code), and the TC Link function allows snapshots or EQ/Dynamics settings stored in the library to be recalled to programmable time code cues.

A time code offset function is also available to offset the time code relative to that fed from an external source. As dynamic automation is such an important feature, touch-sensitive motorized faders are used. This typical high-end approach greatly simplifies writing and modifying the automation data. The DMX-R100 can be switched between two dynamic automation files (A and B) and automation moves can be written in Absolute or Trim modes. In both modes, the automation moves can be rehearsed before overwriting the previous mix in memory. These high-performance automation features make the DMX-R100 very suitable for complex music and audio post mixing that requires extensive auditioning and scene changes.

Inputs/Outputs



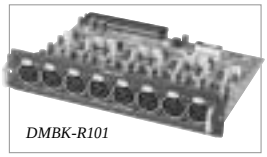
Equipped with 24 analog inputs as well as 8 Aux returns (4 mono analog and 4 mono digital) and 2-track inputs (digital/analog). Analog inputs 1-12 have XLR (for mic inputs and and include 48v phantom power) and 1/4″ TRS connectors (for line level signals). Inputs 13-24 feature neutrik combo connectors (XLR or a 1/4″ TRS. Outputs include stereo program (analog and AES/EBU), Aux Send (8 analog, 2 AES/EBU), control room monitor (6 analog) and studio monitor (2 analog).

DMX-R100

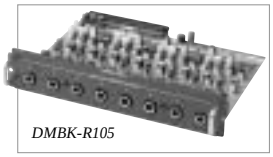
Optional Expansion Boards

For input/output expansion, the DMX-R100 has four option board slots. There are eight optional input/output boards, seven of which allow the user to flexibly expand input/outputs in groups of eight. The eighth optional board, the DMBK-R109 MADI board, allows the addition of 48 inputs/48 outputs via a single board slot. By installing the MADI board in one slot* and the appropriate boards in the three remaining slots, an incredible 72 inputs/outputs become available through the option board slots. As detailed in the chart, the DMX-R100 also interfaces to most popular audio recorders using the appropriate input/output board.

**The DMBK-R109 can only be installed in 'slot-4' of the DMX-R100.*



DMBK-R101



DMBK-R105

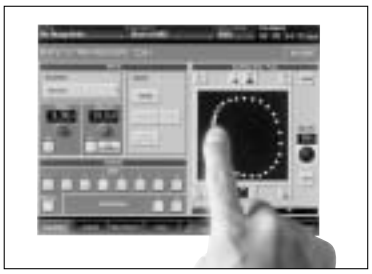
DMBK-R107

Model Number/Description)	Number of Channels	Connector Type
DMBK-R101 (Analog Line-In)	8 inputs	XLR
DMBK-R102 (Analog Line-Out)	8 outputs	XLR
DMBK-R103 (AES/EBU I/O)	8 mono I/O	XLR
DMBK-R104 (Sampling Rate Converter DI)	8 mono ins (x2)	XLR & Optical
DMBK-R105 (Analog Insertion)	8 send/return	1/4″ TRS
DMBK-R106 (ADAT Optical I/O)	8 I/O	Optical x2
DMBK-R107 (TDIF I/O)	8 I/O	D-Sub 25-pin
DMBK-R109 (MADI)	48 I/O	BNC / Optical

Surround Sound Processing

Surround sound is increasingly required for areas such as DVD, film pre-mixing, audio postproduction, etc. As standard, 5.1 surround sound. Furthermore, unique to the DMX-R100, high rate (96 kHz/88.2 kHz) processing is available in 5.1 surround. When high rate processing is selected, the buses and inputs of the DMX-R100 software are reconfigured as follows:

5.1 surround sound at double sample rate	Stereo at double sample rate
24 input channels, 5.1 MTR bus, 2 Aux buses, 4 Aux returns 5.1 CR monitor, 5.1 external source (x 1)	24 input channels, Stereo PGM bus, 4 MTR buses, 2 Aux buses, 4 Aux returns, Stereo CR monitor, Stereo external source, Stereo studio LS



Surround panning is accomplished via the Pan Assign page on the touch screen – by simply tracing the desired panning pattern with a fingertip on the smooth surface of the screen. Alternatively, a PC mouse connected on the back panel of the mixer can be used.

The DMX-R100 has six outputs for the control room monitor, so surround sound monitoring is achieved without sacrificing other outputs and without the need for external monitor switching.

SIU-100 System Interface

The optional SIU-100 is designed for greater system flexibility in applications such as broadcast & post production, and the need for expanded I/O in recording and PA scenarios. Used stand-alone or as an expansion for the DMX-R100, the 5U rackmount SIU-100 accommodates up to eight DMX-R100 style expansion boards, offering a variety of analog and digital inputs and outputs, and high quality microphone preamps.

The SIU-100 interfaces with the DMX-R100 or DAW via the MADI interface, allowing you to position the two units up to 300 meters apart. The Ethernet control capability, gives you the freedom to install the SIU-100 as desired, while also keeping settings at your fingertips.

Additional Features

- No cooling fans are used in the DMX-R100 so it does not generate any acoustic noise.
- Equipped with separate word and video reference input connectors, as well as its own internal reference generator. It also provides multiple machine control with its two 9-pin and MIDI ports. This means that the DMX-R100 can be integrated into any multimedia facility without the need for external synchronization equipment.
- Two DMX-R100's can be cascaded (connected) using the optional DMBK-R109 MADI boards, creating a 96-channel mixer. Since 48 faders become available, this reduces the need for resorting to paging channel faders. Cascade connection at double sample (96kHz/88.2 kHz) rate is also supported.
- The PGM/MTR/AUX buses each have an internal switch to select the bus linkange status (Linked or not) between the master mixer and sub mixer on an individual basis. The AFL bus and solo logic of the master mixer are shared between the master and sub mixer, emulating the operation of a single 96-channel mixer.
- Channel Scribble function allows the entry of names for individual input sources, which then appear on the touch screen next to the channel number. A maximum of seven characters can be entered for each input, and the scribble is displayed in the Channel, EQ and Dynamics GUI screens.

DRE-S777

Sampling Digital Reverberator

The DRE-S777 is a digital reverb device that bridges the gap between "artificial" (synthesizer technology) and "real" to provide a new set of creative tools based on natural-sounding reverberation. This provides audio professionals a totally new approach to sound processing. The DRE-S777 is a technologically advanced digital effects processor that recreates the natural reverberation of concert halls, theaters and sound stages with unparalleled depth, presence and richness. It achieves this breathtaking naturalism by using highly advanced processing that allows audio signals to be combined with sampled data taken from actual acoustic environments. Sony engineers have painstakingly collected sample data from some of the world's most highly regarded concert halls, studios and other acoustic environments, and also from classic analog plate reverb units.The DRE-S777 is supplied with a standard set of sample data, with additional libraries of samples also available.

FEATURES

- Supplied sample CD-ROM contains seven standard presets including data from two different types of concert hall, a recording studio and two churches, plus data from two types of plate reverberator. The DRE-S777 provides for adjustment of parameters such as reverb time, effect/dry balance, EQ and pre-delay. The optional DASK sample discs contain a range of acoustic environments from Europe, America and Japan.
- Nine user caches for quick reverb-program recall.
- The DRE-S777 is capable of a 'self sampling' function, enabling it to capture any acoustic environment. This function requires the optional DASK-704 software module as well as a microphone and speaker system.
- MIDI control
- In addition to stereo reverb modes, it supports multi-channel reverberation. Therefore it is highly suitable for a wide range of multi-channel production tasks, including audio-for-video, and for TV and film post-production applications.
- With the optional DABK-S703 Expansion DSP Board, the DRE-S777 provides a mono input, four-channel output reverb mode. Larger surround sound arrays, for example, 5.1 channel surround, can be provided by using two or more DRE-S777s. The optional DASK software includes reverberation data actually sampled using a five-position mic array, allowing five-channel surround sound effects to be created.
- Reverb mode:
  - Mono in Stereo out (standard)
  - Stereo in Stereo out (with DABK-S703)
  - Mono in 4-channel out (with DABK-S703)
  - Two Mono split in 4-channel out (with DABK-S703)
  - Mono in stereo out at 2Fs (with DABK-S703)
- Direct/Rev: Direct+Reverb/Reverb
- Variable reverb time (0.3-5.5 s max.)
- Sony DSP convolutes an amazing 256,000 sampling points resulting in reverberation that accurately recreates all of the detail of the early reflections and the complexity of the reverberant tails—not only a reverb effect processor, but an ambience emulator!
- Optional DABK-S703 Board upgrades the DRE-S777 for 96kHz sampling, making it compatible with DVD video and audio.
- Optional DABK-S701 (A/D) and the DABK-702 (D/A) converter boards further extend the versatility of the DRE-S777.
- Mixer functions (peak hold, bypass, mixture of dry/wet, muting)
- Four-band parametric EQ
- Factory-presets provided in Memory Stick, as well as set-up data for 92 user-presets
- Operates via jog dial and 4 function keys

Real-World Reverb

To achieve exceptionally natural sounding reverb, it has been necessary to capture the unique "sound" of many different environments; concert halls, churches, studios and so on. The gathering of this data was no mere mechanical process. Rather, it was a series of individual recording projects, each requiring a host of creative decisions familiar to audio professionals. Each project required loudspeakers to radiate the test signals and a microphone array to capture the reverb signature. For each of the locations, multiple samples were recorded using different combinations of loudspeaker and microphone positions. These samples were then combined in the DRE-S777 to provide an extensive range of stereo and surround reverb modes. Sony hardware engineers worked closely with experienced recording engineers to choose microphone types, their directivity and location. Each acoustic space was sampled across a wide range of conditions, and the data supplied on a CD-ROM for use with the DRE-S777.

TASCAM

DM-24

32 Channel 8 Bus Digital Mixer

The Tascam DM-24 is an affordable 32-channel digital mixer that combines 24-bit audio quality, with highly flexible routing capabilities, two powerful onboard effects processors, comprehensive snapshot and dynamic automation as well as dynamics processing and parametric EQ for every channel. The plethora of onboard analog and digital I/O including: 16 mic/line inputs; three 8-channel TDIF I/Os; and 8-channel ADAT optical I/O; along with the wide range of onboard synchronization and machine control capabilities, makes the DM-24 the ideal companion for professional multitrack recording systems such as Tascam's MX-2424 24 track hard disk recorder. The DM-24's 24-bit A-to-D and D-to-A converters and custom 32-bit floating point processor ensures true 24-bit performance throughout the digital signal path.

FEATURES

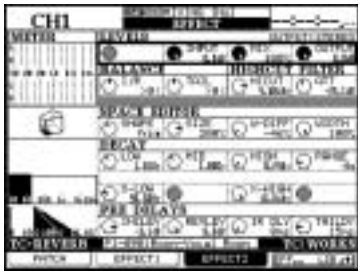
Console Overview

- 32-channel, 8 buss digital mixing console with 6 aux sends at 44.1 or 48 kHz sample rates (16x8x2 at 88.2 or 96 kHz)
- 16 XLR mic, 1/4″ TRS line inputs with inserts and switchable phantom power.
- 24 channels of TDIF I/O, 8 channels of ADAT optical I/O – ideal for any multitrack recording system including Tascam's MX-2424 hard disk recording system.
- Any input can be assigned to any output
- 32-bit internal processing resolution and 24-bit A-to-D, D-to-A converters.
- Each of the 32 input channels is equipped with 4-band fully-parametric EQ, six Aux Sends (assignable pre or post fader) and configurable dynamics section.
- Channels 1-16 also feature a high quality Gate/Expander.

Control Surface

- Sixteen long-throw motorized faders, arranged in three switchable layers, controls up to 32 mono inputs, eight buss sends and six aux sends.
- Each of the 32 inputs can be linked as a stereo pair plus their are eight fader groups and eight mute groups available
- Grouping layers provide further flexibility in grouping arrangements.

Two Fully-Editable Internal Effects Processors



- Processor One gives you the choice of using the TC Works reverb, with over 100 presets, or the Antares Mic and Speaker modeler that emulates the distinctive characteristics of a wide variety of classic and modern microphones, using any standard microphone.
- Processor Two is dedicated to Tascam effects including – Chorus, Delay, Pitch Shifting, Phaser, Flanger, Compressor, Guitar compressor, Exciter, De-esser, Gate and Distortion (with amp simulators)
- Effects can be accessed from the Aux sends 1-6 and returned to any of the 32 channels.
- Effects can also be inserted directly on busses 1-8, the stereo buss or onto one of the 32 channels.
- Effect 1 and 2 can also be run in series (chained) allowing you to run a delay into a reverb, for instance.

Aux Sends

- 6 aux sends, configurable Pre or Post, are provided for each channel and can be freely assigned to the two internal effects processors as well as the 4 assignable sends on balanced TRS jacks, and in pairs to the 2 channel AES/EBU and S/PDIF digital outputs for use with external effects.

Monitoring

- Full control room and studio monitoring facilities are provided, along with an integral talkback microphone and master bargraph meters.
- All popular surround formats are supported including 2+2, 3+2 and 5.1.

Internal Automation – No Computer Required

- The Built-in automation system allows full real time control of almost all mix parameters without the need for connection to an external sequencer.
- Up to eight banks of automation mix files, each containing up to 8000 events, can be stored internally. Up to four banks or 32,000 events can be used per mix.
- Automation is triggered by external MTC, SMPTE time code or from the internal MTC generator.
- You can store and recall 95 user snapshots of the current mixer settings.

DM-24

Input and Master Sections

A. Sixteen XLR mic and 1/4″ TRS line inputs with variable gain and 1/4″ TRS inserts for each channel (phantom power for the mic inputs are switchable in groups of four).

B. Master Section:

– 1/4″ TRS Control Room outputs and Studio Monitor outputs plus two headphone outputs

– 2 Track Input on RCA Connectors

– XLR Main stereo outputs with 1/4″ TRS insert

– Four 1/4″ TRS Sends and 1/4″ TRS Returns can be configured as inserts for tape returns.

Control Surface

C. Each of the sixteen channel strips features a long-throw motorized fader, as well as backlit Record Enable, Channel Select and Mute/Solo buttons and an overload/status LED. The channel strips are arranged in three switchable layers, allowing them to control each of the 32 mono inputs, 8 buss sends and 6 aux sends.

D. Control Room Section:

– Solo and headphone volume controls.

– A volume control, dim switch and mute switch are provided for the control room outputs.

– Three programmable source select switches allow you to designate the desired monitor source.

– 2 x 12-segment LED meters

– The Jog/Data wheel can be used for changing parameter values as well as transport control.

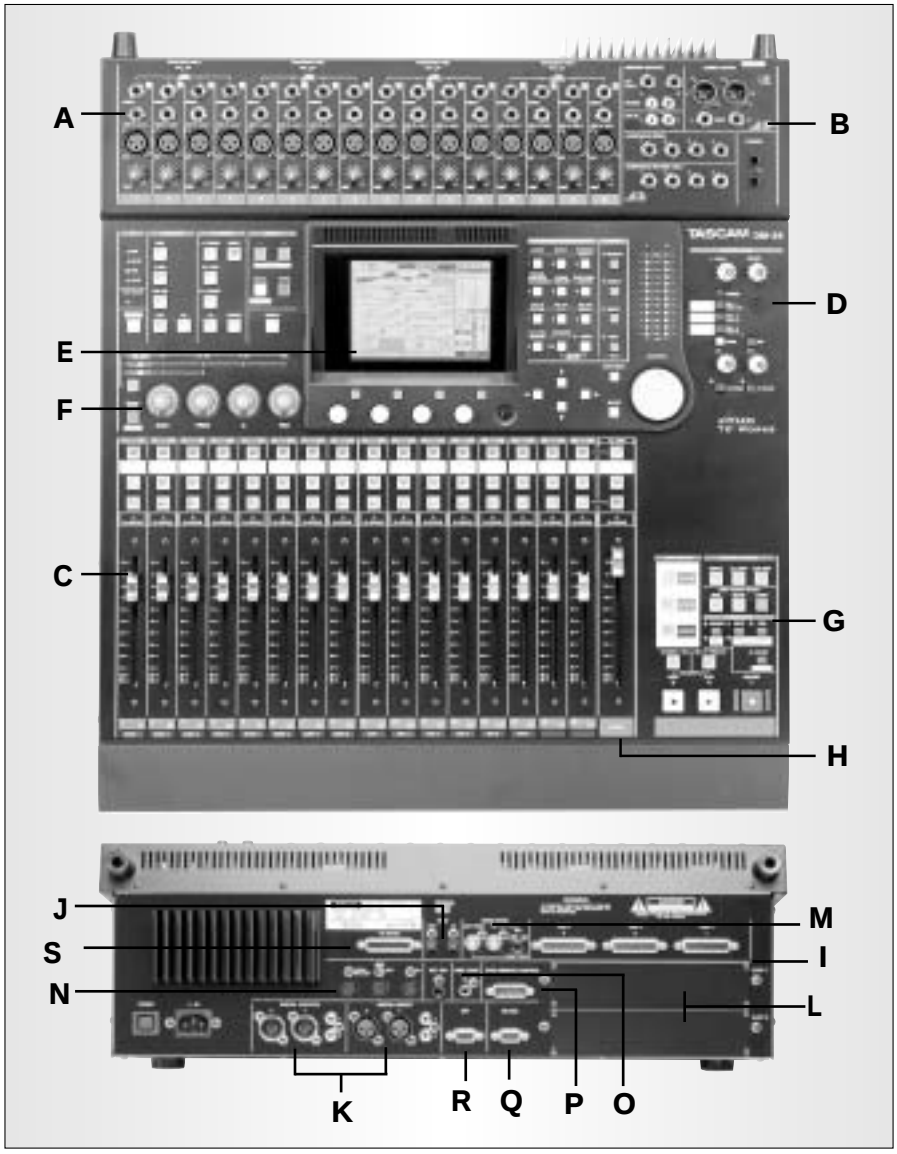
– Built-in talkback mic with level control.

E. The backlit 320 x 240 LCD displays routing assignments, EQ curves, effects parameters, automation data, external control status, synchronization and more. 4 multifunction rotary "POD" controls, located directly underneath the display, provide control over effects, EQ, dynamics, and surround panning parameters and more. The POD controls also transmit MIDI controller data.

F. Four Ring Encoders give instant visual feedback of key EQ, pan and aux send settings.

G. The Transport Section can control a wide variety of external devices using the onboard RS-422, DTRS remote, MMC and GPI ports as well as ADAT Sync by using the optional IF-AD-DM card. Functions include: Play, Record, Stop, FF, REW transport controls; numeric keypad for manual and direct locate; auto punch keys; repeat; memo; edit keys and track arming keys.

H. The dedicated motorized long-throw Master fader is provided for controlling the stereo out buss.



I. Three TDIF I/O interfaces provide 24 channels of buss outs and returns for use with TDIF compatible DAWs and DTRS machines.

J. Eight channels of ADAT Optical I/O provide lightpipe compatible buss outs and returns.

K. Two stereo AES/EBU and two stereo S/PDIF interfaces.

L. Two option slots accommodate additional 8-channel analog and digital interface modules or a cascade module for connecting two DM-24s.

M. Word Sync In and Out/Thru ports (BNC)

N. MIDI In, Out and Thru/MTC Out

O. SMPTE Time Code input (RCA)

P. 15-pin D-sub DTRS compatible Remote output

Q. RS-422 (for Sony 9-pin) 9-pin female D-sub connector

R. GPI 9-pin female D-sub connector allows remote machine start

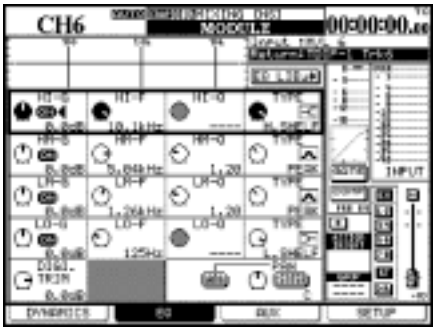
S. 25-pin D-sub connector for use with the optional MU-24/DM Meter Bridge.

TASCAM

DM-24

EQ AND DYNAMICS ON ALL 32 INPUT CHANNELS

4 Band Parametric EQ



High Filter On/Off switchable
Type: Hi-shelving, Peak, LPF
Frequency: 31Hz to 19kHz
Q: 0.27 to 8.65
Boost/Cut: ±18dB, 0.5dB resolution

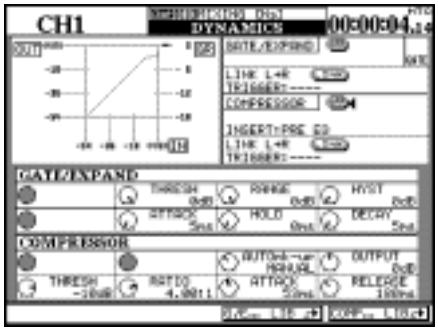
Hi Mid Filter On/Off switchable
Type: Peak, Notch
Frequency: 31Hz to 19kHz
Q: 0.27 to 8.65
Boost/Cut: ±18dB, 0.5dB resolution

Lo Mid Filter On/Off switchable
Type: Peak, Notch
Frequency: 31Hz to 19kHz
Q: 0.27 to 8.65
Boost/Cut: ±18dB, 0.5dB resolution

Low Filter On/Off switchable
Type: Low-shelving, Peak, HPF
Frequency: 31Hz to 19kHz
Q: 0.27 to 8.65
Boost/Cut: ±18dB, 0.5dB resolution

All filters are fitted with "gain flat" switches

Dynamics



Gate:

Threshold	-80dB to 0dB in 1dB steps
Range	60dB to 0dB in 1dB steps
Hysteresis	0dB to 24dB in 1dB steps
Attack time	0ms to 125ms in 1ms steps
Hold time	0ms to 990ms in 100 steps
Decay time	50ms to 5.0s

Expander:

Threshold	–48dB to 0dB, in 1dB steps.
Ratio Values	1:1, 1:2, 1:4, 1:8, 1:16, 1:32, 1:64
Attack	0ms to 125ms in 1ms steps
Release	5ms to 5.00 seconds

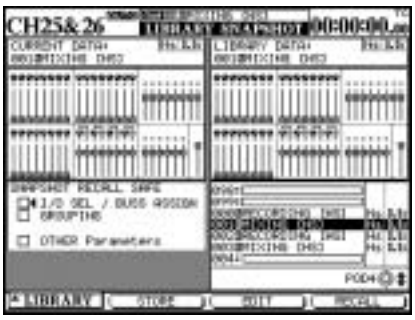
Compression:

Threshold	-48dB to 0dB in 1dB steps.
Ratio	1:1 to ∞:1
Attack time	0ms to 125ms in 1ms steps.
Release time	5ms to 5.0s in 100 steps.
Auto (gain) make-up	switchable
Output gain	-20dB to +20dB in 1 dB steps

Libraries

- The DM-24 allows allows you to store customized settings in libraries. Most of the libraries included a number of locations reserved for read-only presets containing useful points of reference.
- The Library undo/redo function allows a recently recalled library setting to be compared with a previously-loaded setting.

Library Presets	Total	Read-only
Snapshots	100	4
EQ	128	20
Effect 1 (P1)	128	128
Effect 2 (P2)	128	128
User effect library	128	0
Compressor	128	18
Gate/Expander	128	2
Automation	7	–



The snapshot library screen includes a picture of the faders, together with pan/balance settings and mute settings.

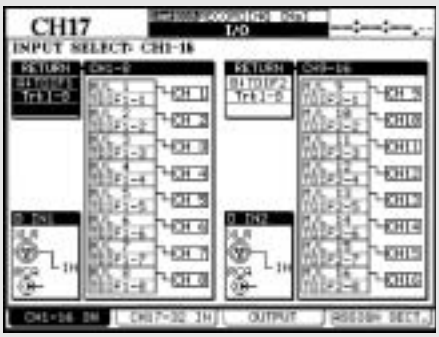
Signal Flow Examples

Tracking:

- Channels 1-16 can be accessed from the 16 mic/line inputs and then assigned to 16 direct outs while channels 17-32 can provide 16 tape returns for monitoring.
- Aux sends 1-6 can be used to create 6 mono or 3 stereo cue mixes.

Overdubbing:

- TDIF returns 1-16 can be assigned to channels 17-32 while channels 1-16 are used for four mic/line inputs, two internal effects returns, and the returns from the third return group. This gives allows you to record a vocal track dry, yet monitor the vocal with reverb in the cue mix.



48 Channel Mixdown:

- The first 24 of the 32 input channels can be accessed by the three TDIF returns.
- The additional 8 input channels are accessed by the two internal effects, a digital stereo pair, and an external send/return.
- The mic/line inputs can also be used, by their direct assignment to Aux 1-2. This means that up to 16 channels of live MIDI sequenced instruments can be added to the 32 input channels.

DM-24

Synchronization and Machine Control

- The DM-24 can be easily configured as a remote controller for a wide variety of devices using the DTRS remote, P2 and MMC protocols. Multiple devices can even be controlled simultaneously.
- MIDI In, Out and Thru/MTC Out allows you to Send MMC, Lock to incoming MTC, Update firmware via Standard MIDI File, offload and upload library and automation data as well as Send/Receive MIDI program changes and MIDI controller data.
- Built-in Mackie HUI emulation provides MIDI-based control surface support for a variety of DAW applications including ProTools, Logic, DP, Cubase and Nuendo.

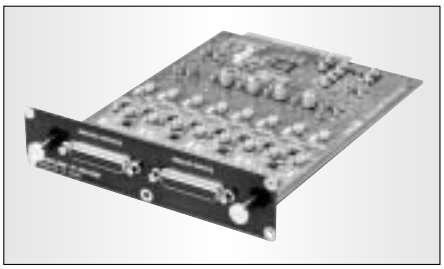
Cascading Two DM-24s

- Two DM-24s can be cascaded together to act as one large console. This requires the IF-CS/DM cascade option card in each mixer.
- Cascading allows the two mixers to share busses 1-8, aux sends 1-6, the solo buss and the stereo buss. Connections to the control room and stereo outputs would be made on the cascade master.
- An AES/EBU option card added to the cascade master can be used for sending the 5.1 mix digitally to your stem recorder while an analog option card could be added to the cascade slave to send the 5.1 mix to the surround monitoring system.

Cascading two DM-24s provides:

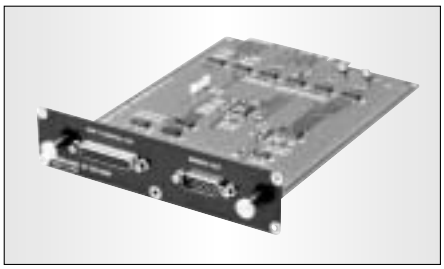
- 64 channels with 32 mic pres
- 48 channels of TDIF I/O
- 16 channels of ADAT lightpipe I/O
- 4 AES/EBU 2 channel digital I/O
- 4 S/PDIF 2 channel digital I/O
- 8 Assignable sends and returns
- 4 Internal effects processors (2 in each)
- 33 Touch sensitive, motorized faders
- The ability to run a 24 track, 24-bit, 96kHz, 5.1 mixing environment.

IF-AN/DM



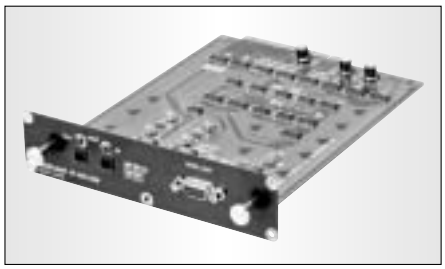
Eight channels of analog I/O on D-25 connectors. Installing two of these cards will give you a total of 32 simultaneous analog inputs.

IF-TD/DM



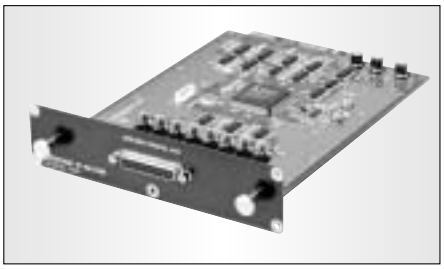
Eight channel TDIF card. The IF-TD/DM is not available for input. It is designed to be used to send the 6 buss outputs to a TDIF stem recorder like a DTRS machine.

IF-AD/DM



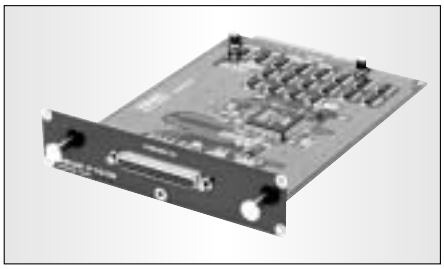
Eight channels of ADAT optical I/O with ADAT Sync. Adding two of these cards will allow you to mix 24 tracks of ADAT. The IF-AD/DM does NOT support 96kHz.

IF-AN/DM



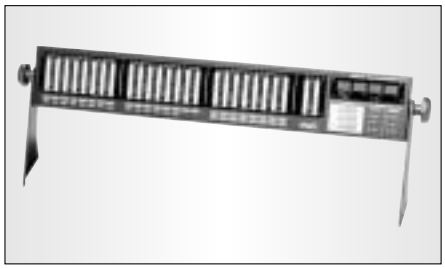
Eight channels of AES/EBU I/O on D-25 connectors. Adding two IF-AE/DM AES/EBU cards provides a total of 20 channels of AES/EBU I/O (four channels come standard on the DM-24.) The IFAE/ DM AES/EBU card supports 4 channels of DUAL LINE or HI SPEED I/O

IF-CS/DM



For cascading two DM-24 digital mixers. To cascade two DM-24s, you must put one cascade card in “SLOT 1” of each DM-24. A cascade cable is included.

MU-24/DM

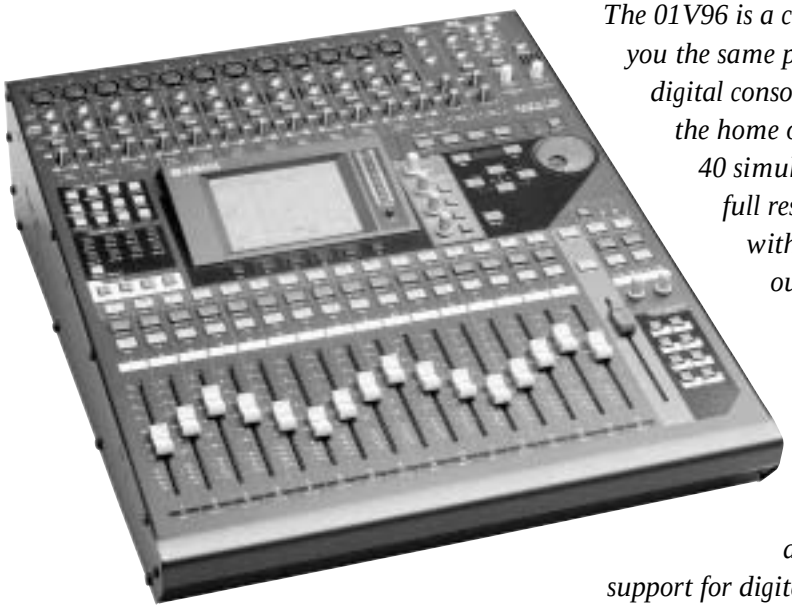


The optional MU-24/DM Meter Bridge provides channel and master metering facilities through LED bargraph displays which are switchable in “layers”.

YAMAHA

01V96

32-Channel 24-bit / 96kHz Digital Mixing Console



The 01V96 is a cutting edge digital mixing console designed to bring you the same performance and reliability of Yamaha's most advanced digital consoles, in a smaller, more affordable format that's ideal for the home or smaller professional production studio. It featuring 40 simultaneous mixing channels (when fully expanded) with full resolution 24-bit/96 kHz audio, a range of stereo effects with 32-bit internal processing and full automation. Right out of the box, the 01V96 provides 24 analog inputs and an 8 channel ADAT optical I/O interface. For further expansion, a single I/O slot will accept Mini-YGDAI digital and analog I/O cards, including third party cards, providing up to 16 inputs and outputs at 48 kHz (8 at 96kHz). The comprehensive control surface allows analog-style hands-on operation along with the added flexibility of providing extensive support for digital audio workstations including ProTools, Nuendo and other popular computer-based recording software.

FEATURES

- 36 simultaneous inputs, 8 busses and 8 auxes, at 24 bit, 96kHZ operation
- Supports 44.1 kHz, 48 kHz, 88.2 kHz, or 96 kHz sample rates
- 24-bit A-to-D and D-to-A converters

24 Analog and Digital Inputs

- 16 analog channel inputs:
  - Channels 1 - 12 will accept balanced XLR microphone signals via high-performance head amplifiers, as well as (bal/unbal) line-level signals via 1/4″ TRS connectors. 48-volt phantom power (switchable in 4-channel groups), trim controls and pad switches are also provided as well as unbalanced 1/4″ TRS inserts that will allow you to insert external processing gear into these channels (pre-A-to-D).
  - Channels 13 - 16 feature 1/4″ TRS (bal/unbal) line inputs with trim controls, that can be configured as four individual inputs or two stereo pairs.
- Eight digital channel inputs (and outputs) are provided via a built-in ADAT optical interface.

Expandable To 40 Inputs

- When you need more I/O or need to configure a system based around a specific protocol, additional I/O can be added via a range of user-installable 8 and 16 channel Mini-YGDAI expansion cards that can simply be plugged into the expansion slot to provide in a variety of formats including ADAT, AES/EBU, TDIF or analog.

All Input Channels Feature

- Independent compressor/limiter and gating/ducking processors on each channel provide complete dynamics control.
- Each channel also features 4-band fully parametric EQ. Each band is fully sweepable from 20 Hz to 20 kHz, with bandwidth variable from 0.1 to 10 and a ±18dB boost/cut range.
- The channel delays have a maximum delay of 452 milliseconds (96 kHz mode). Even the stereo bus, eight mix buses, and eight aux buses have individual compression and EQ!

17 Physical 100mm Motorized Faders

- There are sixteen channel strips each of which features a precision 100mm motorized fader, channel ON/OFF (mute) key, a SOLO key, and a SEL key. The SEL key allows you to assign the channel to the console's Selected Channel section where you have access to detailed controls for panning, EQ and dynamics.
- The 16 channel strips can be instantly switched via four Layer-switching keys to control:
  - Input channels 1 - 16
  - Input channels 17 - 32
  - The Master Layer which controls auxiliary sends 1 - 8 and buses 1 - 8
  - The Remote Layer can be used to control your DAW software including ProTools, Nuendo and more
- Fader Mode keys allow the motor-driven faders to be instantly switched and updated between fader and auxiliary level control.
- The seventeenth motorized fader is reserved for the Master Stereo bus which also features On/Off and Sel keys.

01V96

20-bus Configuration

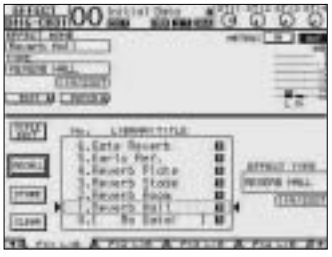
- A wide range of freely configurable signal-routing options are provided to adapt to a variety of tracking, mixing and live sound applications:
  - Main stereo program bus
  - Eight individual mixing buses
  - Two solo buses
  - Eight auxiliary buses

Digital Patching System

- A fast, flexible and easy-to-use digital patching system allows you to assign all of the available inputs, outputs, effects, and channel inserts to any of the console's channels or outputs.
- A direct out function allows the signal from any input channel to be routed directly to any digital or analog output.
- The eight auxiliary buses can also be patched to anywhere in the system.
- The Patch Library allows you to store and instantly recall entire patch setups – ideal for configuring templates for different applications i.e. tracking, mixing, mastering etc.

24-bit Fully Editable Internal Effects

- You can use two stereo effects simultaneously at the 88.2 and 96 kHz sample rates, and up to four stereo effects at 44.1 and 48 kHz.
- Effects can be assigned to an auxiliary bus for send and return operation, or to an insert into any input channel as required.



- An effect libraries with 44 factory and 76 user presets is provided covering a whole host of effects types, including reverbs, delays, modulation-based effects, combination effects, and multichannel effects designed especially for use with surround sound.

Snapshot and Dynamic Automation

- The 99 Scene Memory allows complete console setups to be memorized and instantly recalled via the dedicated SCENE MEMORY controls or via MIDI program changes.
- An external sequencer can be used to record real-time movement of the motorized faders, EQ settings and other parameters, allowing complete mix automation.

Integrated DAW Control

- Extensive support, including full control of mixing and processing parameters, as well as transport/track-arming control and access to editing functions, is provided for Digidesign's Pro Tools system as well as Steinberg's Nuendo DAW. There's also a "General DAW" mode that provides compatibility with other workstations.

Cascade Link

When you really need high capacity – particularly for sound reinforcement applications – the 01V96 offers "01V96 Cascade Link" capability that allows two 01V96 units to be cascaded to create up to an 80-channel mixing system at an unbelievably affordable price!

Rear Panel

- XLR-balanced analog main stereo outputs and 1/4″ TRS (bal/unbal) monitor outputs
- Four 1/4″ TRS (bal/unbal) "Omni" outputs.
- ADAT optical In and Out connectors
- 24-bit coaxial S/PDIF digital 2-track inputs and outputs with on-board sample rate conversion that allows CD players and other digital sources connected to the digital input to be monitored or routed to an input channel without having to be synchronized to the system clock.

Sync and Control Options

- Word clock inputs and outputs (BNC)
- MIDI In, Out and Thru connectors and a USB "To Host" connector are provided allowing computer control via the supplied Studio Monitor software as well as automation via MIDI cc data.
- An expansion slot which will accept a wide range of Yamaha mini-YGDAI expansion cards that can add up to 16 additional channels in a variety of formats.

Expandable Data Libraries

- A number of Data Libraries, each containing an extensive selection of presets, are provided for effects, compression, gating, EQ, I/O patching, and channel setups.
- Presets can be selected and used unmodified, or edited to suit specific requirements. Modified setups can be renamed and saved to the libraries and instantly recalled when needed.

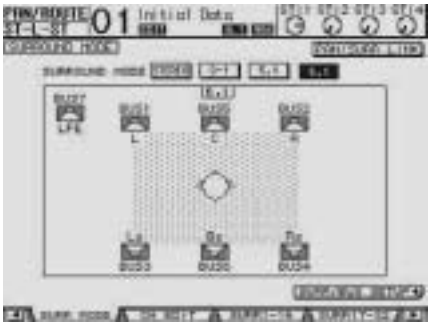
Libraries Presets	Factory	User
Effect libraries	44	76
Compressor libraries	36	92
Gate libraries	4	124
EQ libraries	40	160
Channel libraries	2	127
Input patch libraries	1	32
Output patch libraries	1	32

01V96

Navigation

- The high-resolution 320 x 240 dot LCD panel provides easy visual access to all of the consoles functions and parameters.
- Many of the parameters, such as EQ curves and compression parameters, are displayed graphically so you can see what's happening at a glance.
- Display Access keys determine which type of data will be shown on the LCD panel – a total of 12 selectable categories. This approach minimizes the need to scroll through on-screen lists when you need access to a particular type of data.
- The Selected Channel controls include the hands-on panning and EQ controls for the currently selected channel. Analog-style buttons and knobs provide direct and easy access to the parameters.
- Eight User Defined Keys can be assigned to control any functions you choose. For example, you can use them to recall input patch setups, to arm MTR tracks for recording, or to handle locator functions.
- When the REMOTE layer is selected, the USER DEFINED KEYS are automatically assigned to Pro Tools control functions by default.
- Large cursor, INC/DEC, and enter keys are complemented by a data entry wheel that lets you spin in values quickly and easily.
- The data entry wheel can also be used as a shuttle/ scrub dial for a recorder or DAW.

Surround Panning



Surround is becoming an important part of modern sound production. The 01V96 features 6.1, 5.1 and 3-1 surround panning modes so you can create surround mixes without having to sacrifice features or performance in other areas.

STUDIO MANAGER SOFTWARE (MAC/PC)

SUPPLIED WITH BOTH THE 01V96 AND 02R96

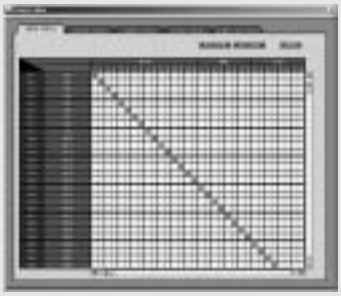
Studio Manager gives you complete access to all parameters for either on-line or off-line control, and the program's visual interface makes it easy to relate on-screen controls to the corresponding console functions. It can also be used to manage an extensive archive of mix data. The Studio Manager includes the following main display pages:

Console Window

Almost a complete virtual mixer, this display shows the primary channel parameters "in-line" as they would appear on an analog console.

Patch Editor Window

When you add several Mini-YGDAI expansion cards to create your system's I/O configurations, input and output patching can become an issue. The Patch Editor makes patch assignments easy with a matrix type visual interface.



Selected Channel Window

Similar to the SELECTED CHANNEL section on the console, this display includes all parameters for the currently selected channel: level, buss assignments, sends, gate, compressor, delay, etc.

Surround Editor Window

When the console is being used in surround mode the Surround Editor display can facilitate surround positioning of individual tracks.

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EQUIPMENT LEASING AVAILABLE

01V96 • 02R96 EXPANSION BOARDS

Connection with 96-kHz Recorders and Workstations

Although the 01V96 handles 96 kHz audio as standard, most of the currently available digital recorders and workstations can handle 96 kHz audio only in double channel mode (using two tracks to make one). In this configuration the 01V96 uses one channel per (96-kHz) track, but twice the number of I/O connections must be used. MY8-AT/TD/AE cards work in double channel mode to handle 16 channels of 44.1/48-kHz audio or up to 8 channels of 96 kHz audio in double channel mode. With the latest equipment that handles 96-kHz audio as standard (in double speed mode like the 01V96) you can make standard connections using the MY8-AE96 card. The MY8-AE96 card can work either in double speed or double channel mode.

MY-16TD

16 channel TDIF format I/O

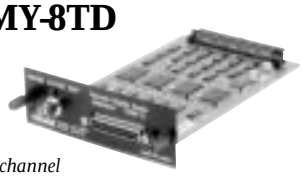
MY-16AT

16 channel ADAT format I/O

MY-16AE

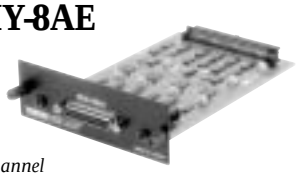
16 channel AES/EBU format I/O

MY-8TD



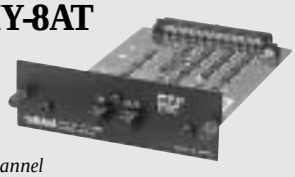
8 channel ADAT format I/O

MY-8AE



8 channel AES/EBU format I/O

MY-8AT

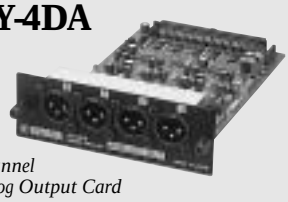


8 channel ADAT format I/O

MY-8mLAN

mLAN interface card

MY-4DA

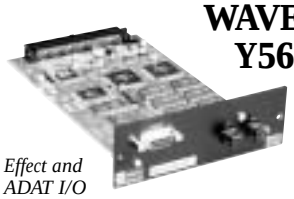


4 channel Analog Output Card

MY-8AE

4 channel Analog Input Card

WAVES Y56K

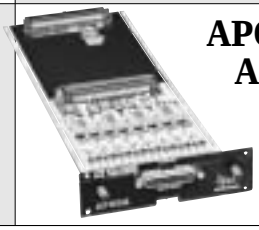


Effect and ADAT I/O

APOGEE AP8AD

A/D Converter

APOGEE AP8DA



D/A Converter

Mfr.	Model	Function	Input	Output*	Format	Resolution	Frequency	Notes
Yamaha	MY16-AT	Digital I/O	16	16	ADAT	24 bit	44.1/48 kHz	2
Yamaha	MY16-AE	Digital I/O	16	16	AES/EBU	24 bit	44.1/48 kHz	2
Yamaha	MY16-TD	Digital I/O	16	16	Tascam	24 bit	44.1/48 kHz	2
Yamaha	MY8-AT	Digital I/O	8	8	ADAT	24 bit	44.1/48 kHz	2
Yamaha	MY8-TD	Digital I/O	8	8	Tascam	24-bit	44.1/48 kHz	2
Yamaha	MY8-AE	Digital I/O	8	8	AES/EBU	24 bit	44.1/48 kHz	2
Yamaha	MY4-AD	Analog In	4	-	Analog	24 bit	44.1/48 kHz	
Yamaha	MY4-DA	Analog Out	-	4	Analog	20 bit	44.1/48 kHz	
Yamaha	MY8-AD24	Analog In	8	-	Analog	24 bit	44.1/48 kHz	
Yamaha	MY8-AD96	Analog In	8	-	Analog	24 bit	44.1/48/88.2/96 kHz	
Yamaha	MY8-DA96	Analog Out	-	8	Analog	24 bit	44.1/48/88.2/96 kHz	
Yamaha	MY8-AE96S	Digital I/O	8	8	AES/EBU	24 bit	44.1/48/88.2/96 kHz	3
Yamaha	MY8-AE96	Digital I/O	8	8	AES/EBU	24 bit	44.1/48/88.2/96 kHz	
Yamaha	MY8-mLAN	mLAN Interface	8	8	IEEE1394	24 bit	44.1/48 kHz	Max. 5 nodes
Waves	Y56K	Effect & I/O	8	8	ADAT	24 bit	44.1/48 kHz	
Apogee	AP8AD	Analog In	8	-	Analog	24 bit	44.1/48/88.2/96 kHz	4
Apogee	AP8DA	Analog Out	-	8	Analog	24 bit	44.1/48/88.2/96 kHz	4

***** *Selectable from Stereo/Bus/Aux/Direct Out/Insert Out/Cascade Out (STEREO, BUS1-8, AUX1-8, SOLO). Details depend on each interface card.*

2. *Can Handle 24 bit/96 kHz using double channel mode* ***3.*** *Sampling Rate Converter for input* ***4.*** *4ch @fs = 88.2, 96 kHz*

YAMAHA

02R96

24-bit/96kHz Digital Mixing Console

The 02R96 Digital Mixing Console is the long-awaited, and fully revised, successor to the revolutionary 02R — the industries first truly professional, affordable digital console. In fact, the 02R96 packs more than five times the processing power into the same footprint size as the original 02R. This power allows the 02R96 to provide 56 channels of full resolution 24-bit/96 kHz audio, full automation of virtually all console parameters with quarter-frame accuracy, surround monitoring, as well as comprehensive range of 96kHz compatible stereo effects with 32-bit internal processing. Out of the gate, the 02R96 provides 24 analog inputs including 16 mic preamps using the same high quality head amplifiers as Yamaha's top-of the-line DM2000 digital mixing console. Thirty Two channels of I/O can be added via four I/O slots that accept a new range of 24-bit/96kHz-capable Mini-YGDAI digital and analog I/O cards. This allows you to configure the 02R96's I/O to suit your specific production environment.

FEATURES

Sonic Spec

- Transparent, full resolution 24-bit/96 kHz audio, with true 24-bit, 128-times oversampling 96kHz A-to-D and D-to-A converters, 32-bit internal processing and 58-bit accumulators.
- 20 Hz–40 kHz (0.5, –1.5 dB) frequency response at 96 kHz sampling rate.
- 105 dB typical dynamic range (A-to-D Input to Stereo Out).

I/O Architecture

- Up to 56 simultaneous input channels each with direct outputs
- 24 balanced analog inputs including sixteen XLR mic, 1/4″ TRS line inputs featuring high-performance head amplifiers derived from Yamaha's top-of the line DM2000, plus eight 1/4″ TRS line inputs.
- An additional 32 inputs (and outputs) can be derived from four rear panel 24-bit 96kHz compatible mini-YGDAI expansion card slots. These slots accept a wide variety of analog and digital (ADAT optical, TDIF and AES/EBU) plug-in cards that will allow you to create an input/output configuration to perfectly suit your system's needs.
- Each input channel has access to 8 aux sends; independent compression and gating / ducking processors; 4-band parameter EQ and up to 453ms. of channel delay.
- 8 freely assignable 1/4″ TRS Omni outputs.
- Dedicated studio monitor and control room monitor outputs.
- AES/EBU digital I/O and two Coaxial 2-track digital I/Os – A sampling rate converter is provided on the digital ins allowing easy integration of 44.1/48 kHz digital audio equipment such as a CD player.
- XLR and 1/4″ TRS stereo outputs plus two analog 2-track inputs.

Superlative Analog Head Amplifiers & Converters

The head amplifiers are derived from the acclaimed DM2000 - some of the finest analog mic preamps available in any console, anywhere. The on-board 24-bit/96-kHz converters ensure that you get an excellent digital representation of the warm, transparent output from these remarkable mic preamps.

Digital Patching

- All available inputs, outputs, effects, and channel inserts can be assigned to any of the console's channels or outputs via the easy-to-use digital patching system.
- Each of the 56 input channels can be routed to their direct outputs; subgrouped to the eight bus outputs; and to the main stereo outputs – simultaneously if need be.
- The eight aux busses can be patched to anywhere in the system, and patch setups may be stored in the patch library for instant recall at any time.
- Input and output ports can be named for easy identification and patches can be stored in the I/O Patch libraries.

EQ

- 4-band parametric EQ is provided on all Input and Output Channels with a choice of a "type I" EQ algorithm or a newly developed "type II" EQ algorithm.

Dynamics

- Independent compression and gating / ducking processors are provided for all 56 Input Channels. Additionally, compression is provided for each output.

02R96

– Control Surface–

A. Variable gain, signal present and peak LEDs are provided for all of the analog inputs plus independent switchable 26dB pad and phantom power on the first 16 analog inputs as well as on/off switches for the inserts I/O.

B. The 320 x 240 fluorescent backlit LCD display is enhanced by twelve Display Access keys that allow you to instantly switch the display to specific editing, utility and metering functions such as Automix, Channel Groups and Pairs, I/O Patching, MIDI and Remote setups and more.

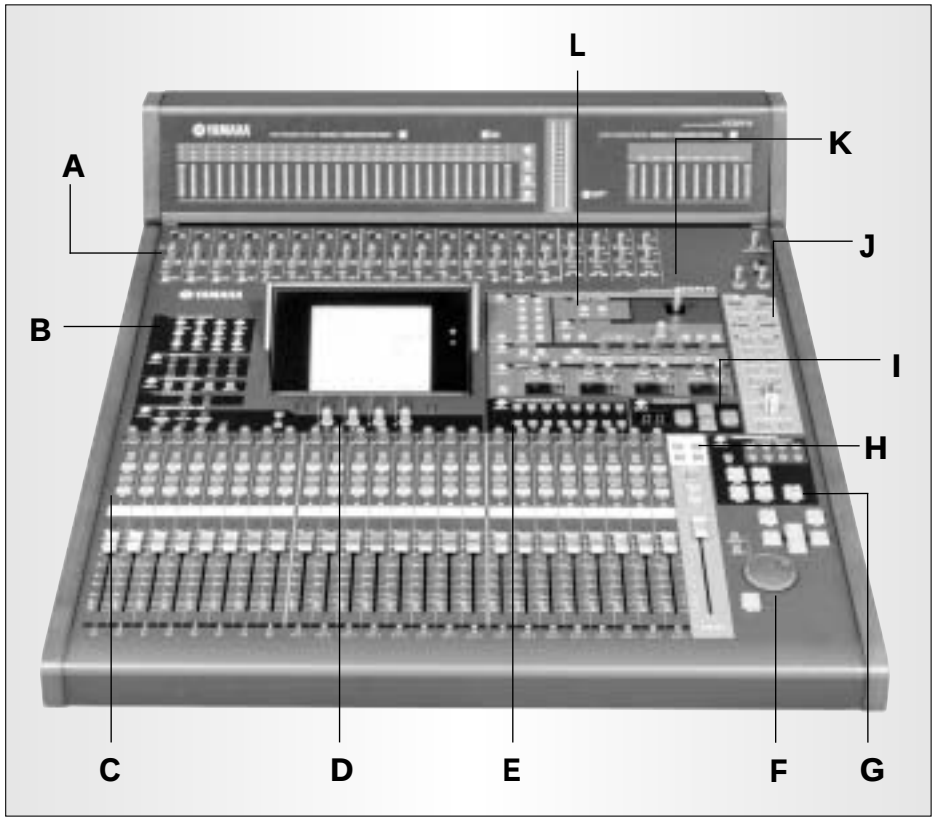
C. Each of the 24 channel strips feature: A touch-sensitive 100 mm motorized fader; a rotary encoder for controlling Pan, Aux Send levels, or user assigned parameters; a channel On/Off key and Solo key; and an Auto key to turn mix automation on or off for that channel. A SEL key allows you to assign a channel strip to be represented on the LCD display and allows you to control the dynamics, EQ, buss assignment and panning for that channel via the dedicated Selected Channel controls.

D. Editing the four internal effects processors is made easy with the parameter up/down selection keys and four rotary encoders. Four keys, located above the encoders switch between menu tabs to reveal additional editing pages.

E. Up to 16 functions from a list of over 150 can be assigned to the User Defined Keys. Setups can be stored in any of the four available banks. Example functions include track arming, scene recall and muting surround monitor outputs.

F. A large Parameter wheel is provided for editing parameter values and scrolling through Scene and library lists. Shuttle and Scrub mode buttons allow you to convert the Parameter wheel to be used for machine control of your DAW or MMC controlled device.

G. Standard transport buttons (Stop, Play, Rec, FF and Rew) as well as as well as 8 locate keys are provided for controlling your DAW via MMC commands.



H. Four Layer switching keys let you access all 56 inputs, the 8 AUX sends and 8 busses as well as a Remote Layer for controlling your DAW via the 24 channel strips.

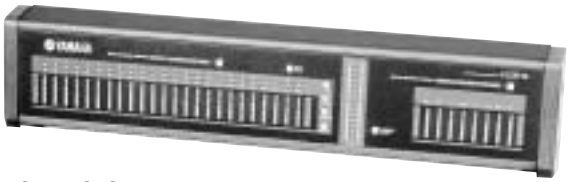
I. A numeric display right next to the Store, Recall, and Up/Down keys shows the current scene number - 01 through 99. Additional scene memories can be managed via a computer running the supplied Studio Manager software.

J. The monitor section features source select keys as well as independent Control Room, Studio, Headphone and Surround level controls. A talkback mic with level control, on/off and Dim switches is also provided.

K. The joystick can be used for surround panning, normal panning as well as parameter control for the 5.1 Reverb effect.

L. Detailed control of dynamics, EQ, buss assignment, panning and surround positioning is available via the Selected Channel controls.

The Optional MB02R96 Meter Bridge Features:



– Twelve 12-segment level meters that can be used to display pre-EQ, pre-fader, or post-fader input channel signal levels.

– An additional eight meters displays the levels for the console's eight busses.

– A 32-segment stereo meter is also provided for the main stereo program.

02R96

– Rear Panel–

A. Sixteen balanced XLR mic and 1/4″ TRS line inputs with 1/4″ TRS inserts (+4dB unbalanced).

B. Channels 17 - 24 feature balanced 1/4″ TRS line level inputs.

C. Analog Master I/O Section

– XLR stereo outputs (+4dB balanced)

– Left and right 1/4″ balanced TRS control room monitor outputs (+4dB balanced)

– Dedicated left and right 1/4″ TRS studio monitor outputs (+4dB balanced)

– 1/4″ TRS studio (+4dB balanced) and RCA (-10dBv unbalanced) 2-track inputs.

– RCA stereo outputs (-10dBv unbalanced)

D. Eight freely assignable balanced 1/4″ TRS Omni outputs can be independently accessed from the Bus Outs, Aux Sends, the Stereo Out, Insert Outs, Direct Outs, or Surround Monitor Channels.

E. AES/EBU digital I/O (XLR) and two coaxial digital I/Os.

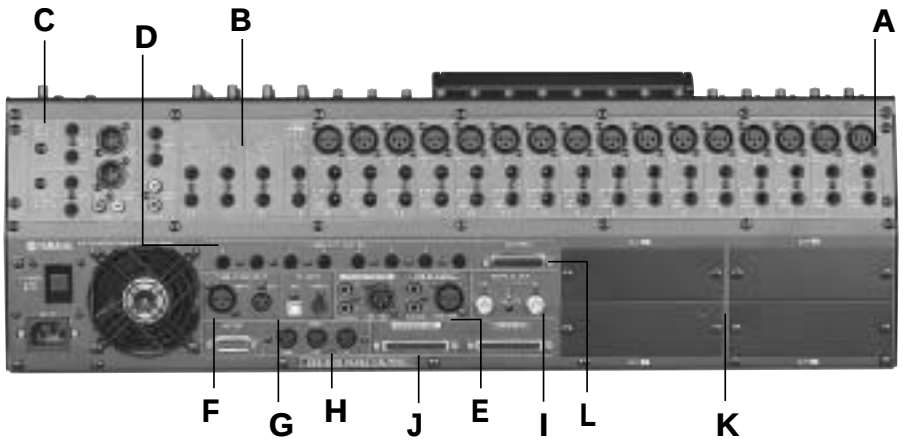
F. A balanced XLR SMPTE timecode input and a dedicated MTC (MIDI Time Code) input are provided for synchronizing the Automix functions with an external device.

G. A USB "To Host" port and 8-pin mini serial DIN "To Host" port allow MIDI communication (including MTC) between the 02R96 and your host computer. The "To Host" ports are ideal for integrating the included Studio Manager Software.

H. MIDI IN, OUT, and THRU ports can be used to send/receive program changes for scene recall, control changes and parameter changes for real-time parameter control, Bulk Dump for data storage, MIDI Clock, MTC, and MMC.

I. Word Clock In and Out is provided by 75Ω BNC connectors – termination can be switched on and off.

J. The 64-pin Cascade In and Out ports can be used to cascade up to four 02R96s to create a multiple-unit mixing system with up to 224 input channels.



K. Four mini-YGDAI slots support up to 32 additional 24-bit/96kHz-capable inputs and outputs (64 I/Os at 44.1/48kHz) via optional analog and digital I/O cards including AES/EBU, ADAT lightpipe, TDIF, and mLAN.

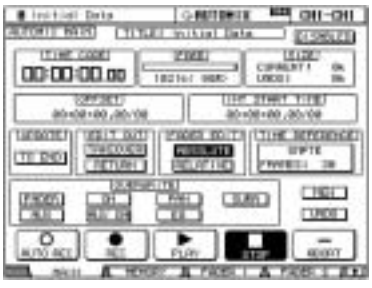
L. A 25-pin D-sub GPI connector allows external equipment to be triggered from specified faders or User Define Keys.

Machine Control

- Both Sony 9-pin (P2) and MMC protocols are supported for external machine control. Control can be switched between MTR and master target machines.
- You can control the transport and locate functions of up to eight external recorders that support MMC (MIDI Machine Control). Machines that support MMC can be controlled by connecting them to the 02R96's MIDI, SERIAL, USB, or with the optional mLAN I/O Card installed in expansion Slot #1.

Automix Functions

- Allows dynamic automation of virtually all mix parameters, including Levels, Mutes, Pan, Surround Pan, Aux Sends, Aux Send Mutes, EQ, effects, and Plug-Ins.
- Events are recorded in real time and can be edited either offline, with high resolution 1/4 frame accuracy, or by re-recording with punch in/out



- The smooth and quiet touch-sensitive, 100mm motorized faders make writing and updating automation fast and intuitive.
- You can specify which parameters will be recorded, and punch channels in and out of recording on-the-fly.
- User Defined Remote Layer operations, and scene and library recall operations can also be automated, combining snap shot and dynamic mix automation
- Automix can be synchronized to an external timecode source or to the internal timecode generator
- Up to 16 Automixes can be stored in the Automix library – They can also be stored to an external MIDI device, such as a MIDI data filer, by using MIDI Bulk Dump

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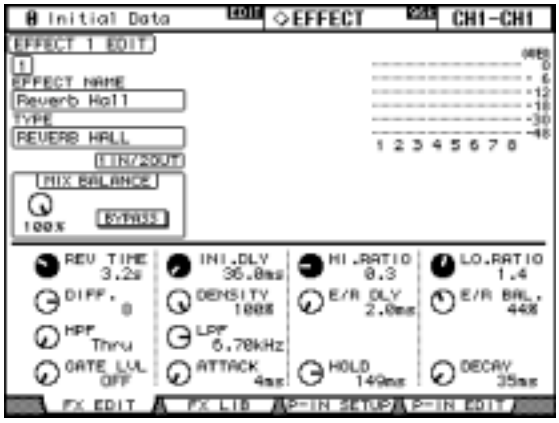
02R96

Four Multi-Effects Processors

- Effects types include reverbs, delays, modulation-based effects, combination effects, and multichannel effects designed especially for use with surround sound.
- Effects processors 2–4 feature assignable stereo inputs and outputs while processor 1 features eight assignable inputs and outputs for use with surround effects.
- An Effects library is provided with 52 preset and 76 user memories.
- Effects processor inputs can be fed from the Aux Sends, Input and Output Channel Insert Outs, or the outputs of another effects processor
- Effects processor outputs can be patched to the Input Channels, Input and Output Channel Insert Ins, or the inputs of another effects processor allowing you to chain processors together in series.
- Joystick control of early reflections and reverb with the Reverb 5.1 effect
- User defined plug-ins for external effects control via MIDI, with Learn function



MULTI-EFFECTS LIBRARY FOR 01V96 • 02R96

Reverbs

#	Preset Name	Description
1	Reverb Hall	Concert hall reverberation simulation with gate
2	Reverb Room	Room reverberation simulation with gate
3	Reverb Stage	Reverb designed for vocals, with gate
4	Reverb Plate	Plate reverb simulation with gate
5	Early Ref.	Early reflections without the subsequent reverb
6	Gate Reverb	Gated early reflections
7	Reverse Gate	Gated reverse early reflections

Delays

#	Preset Name	Description
8	Mono Delay	Simple mono delay
9	Stereo Delay	Simple stereo delay
10	Mod.delay	Simple repeat delay with modulation
11	Delay LCR	3-tap (left, center, right) delay
12	Echo	Stereo delay with crossed left/right feedback

Modulation-based Effects

#	Preset Name	Description
13	Chorus	Chorus
14	Flange	Flanger
15	Symphonic	Proprietary Yamaha effect that produces a richer and more complex modulation than normal chorus
16	Phaser	16-stage stereo phase shifter
17	Auto Pan	Auto-panner
18	Tremolo	Tremolo
19	HQ.Pitch	Mono pitch shifter, producing stable results
20	Dual Pitch	Stereo pitch shifter
21	Rotary	Rotary speaker simulation
22	Ring Mod.	Ring modulator
23	Mod.Filter	Modulated filter

Guitar Effects

#	Preset Name	Description
24	Distortion	Distortion
25	Amp Simulate	Guitar amp simulation

Dynamic Effects

#	Preset Name	Description
26	Dyna.Filter	Dynamically controlled filter
27	Dyna.Flange	Dynamically controlled flanger
28	Dyna.Phaser	Dynamically controlled phase shifter

Combination Effects

#	Preset Name	Description
29	Rev+Chorus	Reverb and chorus in parallel
30	Rev->Chorus	Reverb and chorus in series
31	Rev+Flange	Reverb and flanger in parallel
32	Rev->Flange	Reverb and flanger in series
33	Rev+Sympho.	Reverb and symphonic in parallel
34	Rev->Sympho.	Reverb and symphonic in series
35	Rev->Pan	Reverb and auto-pan in series
36	Delay+ER.	Delay and early reflections in parallel
37	Delay->ER.	Delay and early reflections in series
38	Delay+Rev	Delay and reverb in parallel
39	Delay->Rev	Delay and reverb in series
40	Dist->Delay	Distortion and delay in series

Others

#	Preset Name	Description
41	Multi.Filter	3-band parallel filter (24 dB/octave)
42	Freeze	Simple sampler
43	Stereo Reverb	Stereo reverb
44	Reverb 5.1	6-channel reverb for 5.1 surround
45	Octa Reverb	8-channel reverb for 7.1 surround
46	Auto Pan 5.1	6-channel auto pan for 5.1 surround
47	Chorus 5.1	6-channel chorus for 5.1 surround
48	Flange 5.1	6-channel flanger for 5.1 surround
49	Sympho. 5.1	6-channel symphonic effect for 5.1 surround
50	M. Band Dyna.	Multi-band dynamics processor
51	Comp 5.1	Multi-band compressor for 5.1 surround
52	Compand 5.1	Multi-band compander for 5.1 surround

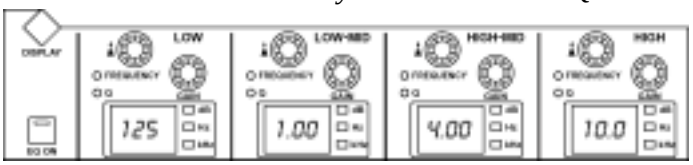
YAMAHA

02R96

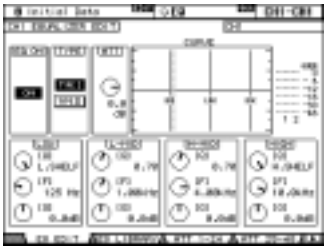
DIGITAL MIXERS

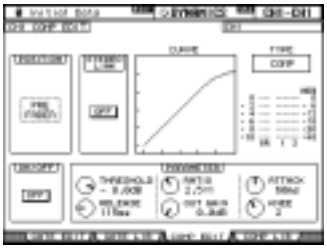
412

Four Band Fully Parametric EQ



Boost/Cut:	±18.0 dB in 0.1 dB steps
Frequency Range:	21.1 Hz to 20.0 kHz (120 steps per 1/12 octave)
Q (Low Band):	HPF, 10.0 to 0.10 (41 steps), L.SHELF
Q (Low-Mid Band):	10.0 to 0.10 (41 steps)
Q (High-Mid Band):	10.0 to 0.10 (41 steps)
Q (High Band):	LPF, 10.0 to 0.10 (41 steps), H.SHELF

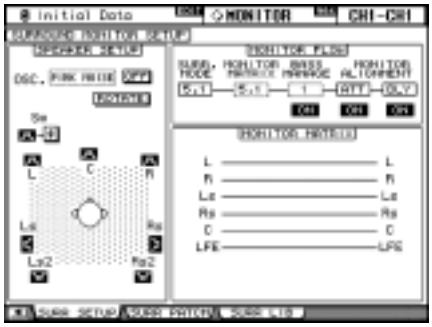




Dynamics

- Independent compression and gating processors are available on all input channels; compression is also provided on all output channels.
- The Selected Channel Dynamics [GATE/COMP] button allows you to switch the dedicated dynamics section to control the Compression or Gating.
- Variable Threshold, Ratio, Attack, Release, and Gain controls are provided for the Compressor.
- Variable Threshold, Range, Attack, Decay, and Hold controls are provided for the Gate.

A Complete Surround Solution



- Everything you need for 5.1 surround processing, panning and monitoring is provided without the need for any external equipment or the need to reassign channels for surround use.
- Multi-channel surround pan/position displays allow you to see where multiple channels sit in the surround mix at a glance.
- The joystick provided for surround panning in the console's SELECTED CHANNEL control section has high 128 x 128 step resolution providing exceptionally smooth control.
- A divergence parameter can be used to adjust the hard/phantom center ratio for each channel.
- The monitor processing section includes individual bus (speaker) muting, attenuation and delay, overall level control for all monitor outputs, stem mix monitoring via independent source selection capability for each expansion slot, bass management for subwoofer delivery, and independent delays for precision speaker tuning.

Remote Control

- Control and manage your 02R96 from your Mac or PC using the bundled Studio Manager software.

Integrated DAW Control

- Mixing and processing parameters, as well as transport control and editing functions of leading digital audio workstations and computer-based recording systems can be controlled directly from the 02R96 control surface. Extensive support is provided for Digidesign ProTools, Steinberg's Nuendo, as well as Emagic's Logic Audio.

Expandable Data Libraries

Setting up EQ, compression, and other parameters for a mix from scratch can be a daunting task, so Yamaha has provided an extensive selection of presets in a range of "libraries" that can simply be selected and used unmodified, or edited to suit specific requirements. Libraries are provided for effects, compression, gating, EQ, I/O patching, and more. Of course, your own setups can be added to the libraries for instant recall whenever they are needed.

- An extensive selection of presets in a range of "libraries" may be selected and used unmodified—or edited to suit specific requirements. Libraries are provided for effects, compression, gating, EQ, I/O patching, and more; user setups can be added to the libraries for instant recall.

	Preset	User
Effect library (Effect 1–4)	52 (Effect 2–4: 44)	76
Presets		
Compressor library	36	92
Gate library	4	124
EQ library	40	160
Channel library	2	127
Surround Monitor library	1	32
Input patch library	1	32
Output patch library	1	32
Bus to Stereo library	1	32

B&H PHOTO - VIDEO - PRO AUDIO

The 5 Ws of mLAN

What is it ?

mLAN is the enabling technology for creating an intelligent, managed local area music network (LAN) using Firewire. mLAN not only carries multi channel digital audio and MIDI over IEEE1394 Firewire, it includes the connection management so you can easily manage your entire network.

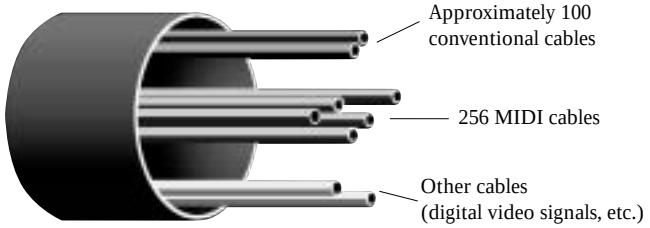
One of the reasons for the complexity and inflexibility of many of today's music production systems and studios is the sheer number of connections involved. All the MIDI and audio signals are routed separately, and each MIDI cable can handle up to 16 channels of data. Audio usually requires a separate cable — whether analog or digital — for each channel of audio for both input and output.
Yamaha's innovative mLAN networking system neatly overcomes all these problems by allowing many channels of digital audio and MIDI music data to be transferred bi–directionally via a single Firewire cable. Extremely powerful music systems can be quickly and easily configured using mLAN-compatible musical instruments, computer components and interface devices. You don't need to worry about the order you connect the devices in, and mLAN ports are "hot pluggable" so you can unplug and plug in devices without having to power-down or reset the system.

When is it?

First generation mLAN products that use the 200 mbps PH-1 chipset are currently available. They allow 8 channels of 48 kHz digital audio i/o and 128 2 Ports (times 16 Channels) of MIDI.
Second generation products are now in development and will be released using the latest PH2 400 mbps chip set. They will be capable of up to 16 x 64 channels of 96 khz audio i/o (or 32 x 128 channels at 44.1 or 48 khz) along with 256 MIDI channels (8 ports times 16 Channels).

Using the software patch bay application provided with all mLAN products, you can easily reconfigure your system – connecting and disconnecting devices as required – without having to physically plug, unplug, or re-route any cables at all.

IEEE 1394 CABLE CONCEPTUAL DIAGRAM



Although computer control can be an advantage in mLAN networks, mLAN allows music and audio devices to be networked without a computer. This allows you to configure an mLAN system for live performances systems.

The other problem that exists in today's market is proprietary formats. Some of these proprietary systems use a Firewire connector, but you can only connect equipment from the manufacturer of the proprietary format which severely limits your choices of gear that you integrate into your studio.

Who is it ?

mLAN was developed by Yamaha, but we realized that for mLAN to be successful, it needed broad participation from other manufacturers. It is available to other manufacturers as a royalty free (no-cost) license. Currently there are over 40 manufacturers who have signed on as mLAN licensees and 8 manufacturers have developed first generation products. Those with current mLAN products include **Apogee**, **Korg**, **Kurzweil**, **Otari**, **Presonus**, **Swissonic**, **Terratec** and **Yamaha**.

Where is it?

mLAN works in Windows XP and there are both ASIO and WDM multi-client drivers available for the PC platform. mLAN is also included in Core Audio as part of OSX 10.2.3 so, you will not even need a driver for Macintosh computers.

Why is it?

So what are the advantages of mLAN over other Firewire audio devices?

Firstly, because other devices use proprietary formats, you can only connect devices from that particular maker to your system. With mLAN, you can hook up an Apogee converter with a Presonus mic pre and a Yamaha synth in one intelligent, well-managed networking system. Because it uses a standard 1394 connection, you can even run video and other data on the same Firewire cable without affecting your mLAN network.

Second, mLAN doesn't require a computer. You can setup your mLAN system at home using a computer and then take the gear to a live gig and hook it together. MLAN will remember the pre-configured setup and re-establish the network and all it's connections.

Third, mLAN handles word clock arbitration so you can run different devices at different sampling frequencies on the same network. Anyone whose tried to setup a digital audio studio knows how important word clock is for a successful studio setup.